COLOUR VIDEO CASSETTE RECORDER

SANYO

VCR TECHNOLOGY

(PAL Beta hi-fi SYSTEM)

Original Model: VTC-M40 series
VTC-M50 series

Beta

Betahi-fi

TABLE OF CONTENTS

1. GENERAL 1
   1. OPERATION DESCRIPTION OF VTC-M40 2
      (1) PART NAMES AND FUNCTION. 2
      (2) BETA hi-fi AUDIO RECORDING 4
      (3) RECORDING LEVEL ADJUSTMENTS 5
   2. OPERATION DESCRIPTION OF VTC-M50 6
      (1) PART NAMES AND FUNCTION. 6
      (2) AUDIO RECORDING LEVEL ADJUSTMENTS . . 9
      (3) RECORDING STEREO/BILINGUAL BROADCASTS 10
      (4) BETA hi-fi AUDIO RECORDING 11
      (5) PLAYBACK STEREO/BILINGUAL BROADCAST TAPE 12
      (6) AUDIO DUBBING 13

2. AUDIO CIRCUIT 15
   1. AUDIO SIGNAL SYSTEM CIRCUITRY 15
      (1) TV PICTURE AND SOUND ARE MADE TO AGREE 15
      (2) AUDIO DUBBING (AFTER RECORDING) OF AUDIO SIGNALS IS POSSIBLE 15
      (3) MUTING. 16
      (4) NOISE REDUCTION. 16
      (5) AUDIO CIRCUITRY CONFIGURATION. 17
   2. AUDIO hi-fi VIDEO 18
      (1) AUDIO hi-fi 18
      (2) OUTLINE OF THE RECORDING SYSTEM 19
      (3) PRINCIPLES OF DEEP RECORDING. 19
   3. PAL BETA hi-fi SYSTEM 22
      (1) PAL (CCIR) SYSTEM AND NTSC SYSTEM 22
      (2) BASIC FORMAT AND PERFORMANCE OF PAL BETA hi-fi SYSTEM 23
      (3) TAPE FORMAT 24
      (4) AUDIO FM CARRIER FREQUENCY 24
      (5) OPTIMUM RECORDING CURRENT 26
      (6) ROTARY TRANSFORMER 27
   4. AUDIO FM CIRCUITRY 28
      (1) OUTLINE AND FEATURES OF AUDIO FM CIRCUITRY 28
      (2) BLOCK 1 29
      (3) BLOCK 2 33
      (4) BLOCK 3 34
      (5) BLOCK 4 38

3. AUDIO FM CIRCUIT 39
   1. OUTLINE 39
   2. AUDIO FM CIRCUIT COMPOSITION. 41
      (1) BLOCK 1: CX20097. 42
      (2) BLOCK 2: CX20104. 45
      (3) BLOCK 3 and 4: CX20105 47
      (4) BLOCK 5: PLAYBACK PREAMPLIFIER AND BANDPASS FILTERS (BPF) 49
   3. DESCRIPTIONS OF THE MAIN AUDIO FM BLOCKS 50
      (1) SNR BLOCK 50
      (2) FREQUENCY MODULATION BLOCK 56
      (3) AUDIO FREQUENCY DEMODULATION BLOCK. 58

4. NORMAL AUDIO AND AUDIO SIGNAL SELECT CIRCUITS 62
   1. OUTLINE 62
   2. DESCRIPTION OF THE ISC IN THE NORMAL AUDIO AND AUDIO SIGNAL SELECT CIRCUITS . . 65
      (1) LA7042 65
      (2) BA5115L 66
      (3) TK15021. 66
      (4) TK15050. 67
      (5) LA3220 67
      (6) TC4066BP 67
      (7) HD14052BP. 68
      (8) TK15090Z. 68

5. VIDEO CIRCUIT 70
   1. OUTLINE 70
   2. DESCRIPTION OF VIDEO CIRCUIT ICS. 72
      (1) LA7007S. 72
      (2) LA7210 72
      (3) LA7020 73
      (4) LA7025 73
      (5) LA7013 74
      (6) TA7788AP. 74
      (7) TA7722P. 75
      (8) TA7357AP. 76
      (9) BA7007 76

SUPPLEMENT (multiplex sound system for VTC-M50) 77
SEPARATE BLOCK DIAGRAM for VTC-M40 (VIDEO, AUDIO)
SEPARATE CIRCUIT DIAGRAM for VTC-M50 (VIDEO, AUDIO)

SANYO ELECTRIC CO., LTD.
SANYO ELECTRIC TRADING CO., LTD.
OSAKA, JAPAN

M4P/M5P

1. GENERAL

This PAL Beta hi-fi Training Manual contains the principles of audio hi-fi concerning the new functions and new circuits for hi-fi sound which have been added to the previous PAL Beta format VCR system in order to improve the sound quality.

The manual uses the VTC-M40 series as a reference model to explain the theories and operating principles of the Beta hi-fi system.

Thus, because the basic operating principles of the Beta hi-fi system will remain the same even for other model (for example, the VTC-M50 series) or even if the IC composition or signal processing is changed, it is important to obtain a thorough understanding of the contents of this Training Manual.

One important function in the Beta hi-fi circuit is the selection of either the conventional audio (also called normal audio) signal or the audio FM signal by the audio signal switching circuit in accordance with the control signals from the various operation switches. This circuit uses digital ICs and discrete components to combine the switch control signals.

We will, therefore, now explain the operation functions of the ICs used in the audio signal switching circuit.

As seen in Table 1, the ICs used in the video circuit in previous models have been changed, primarily in the chroma signal processing system.

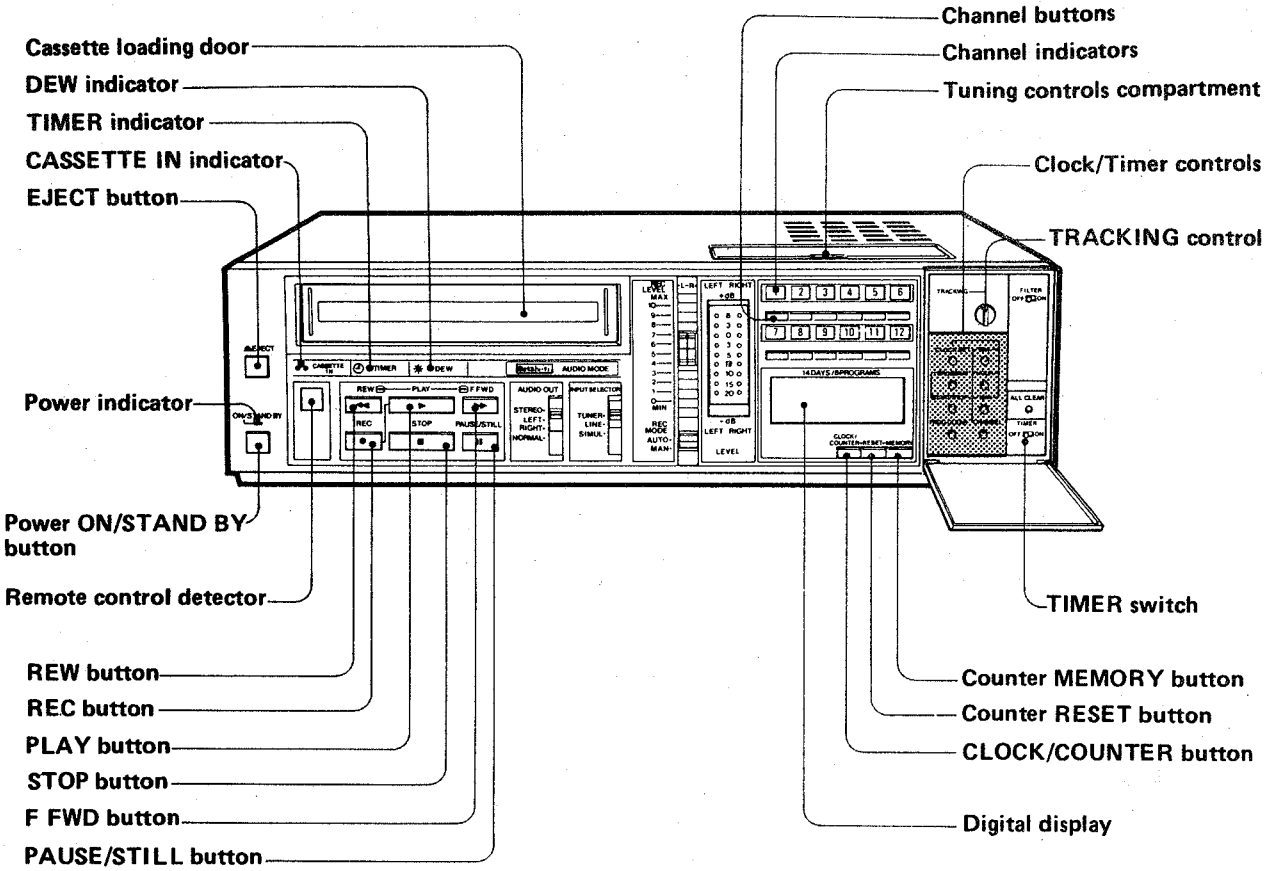
However, because the basic principles of signal processing in the video circuit of PAL format VCRs are almost identical, only the operation specifications of the new ICs are explained here. Also, a brief explanation of the operation and signal flow for each mode is included with the block diagram of the video circuit.

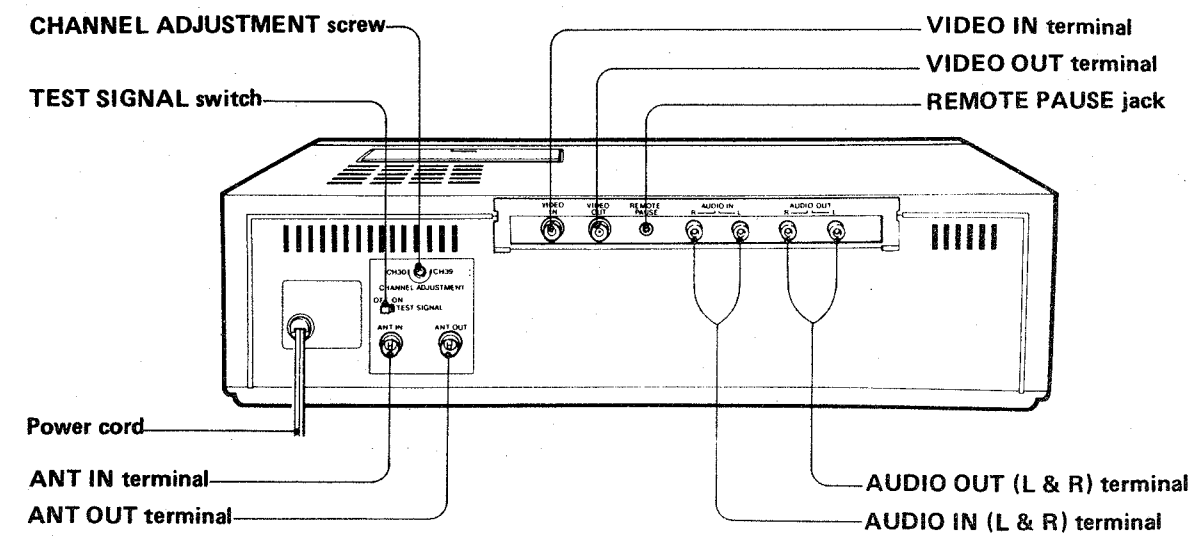
Now we will introduce the functions and uses of the operation buttons related to the basic Beta hi-fi VCR models, the VTC-M40 series and the VTC-M50 series.

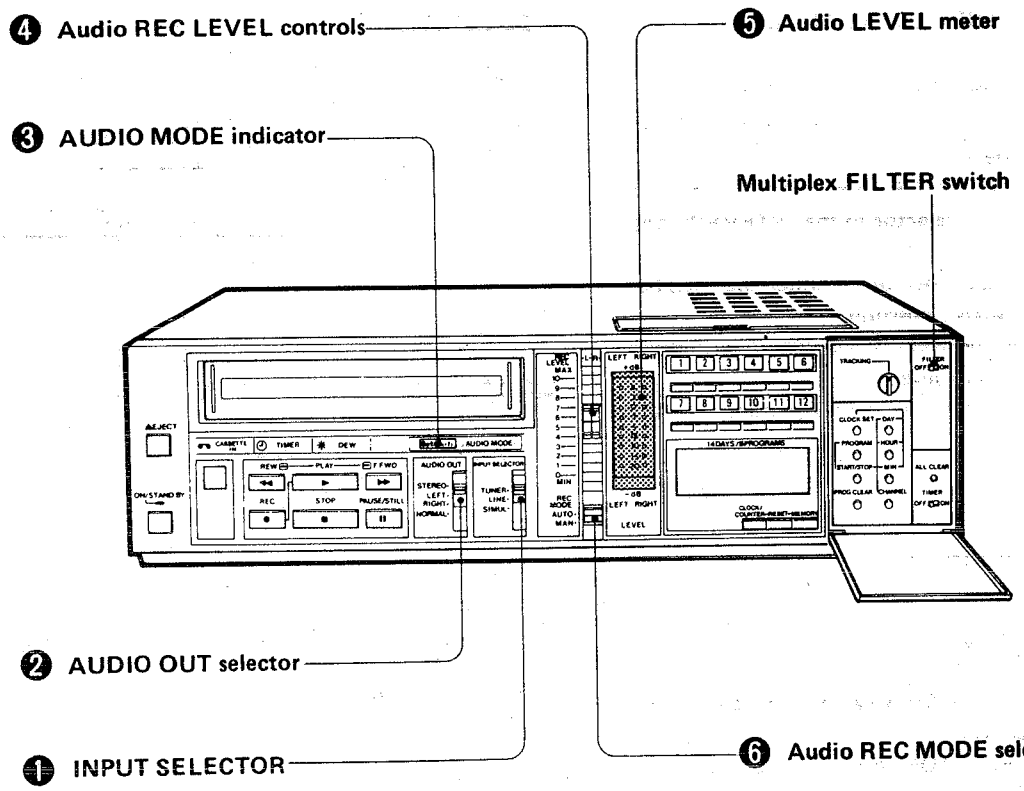
Main IC / Basic model	Luminance (Y) signal process	Chrominance (C) signal process
VTC-9300P series	CX-134A (Pre amp) CX-131A (REC process) CX-135 (PB process)	CX-136A (REC/PB process) CX-137A (APC, AFC process) CX-150 (APC process) CX-145 (sync separator)
VTC-5500P series	CX-134A (Pre amp) CX-187 (REC/PB process) CX-135 (PB process)	CX-136A (PB process) CX-832 (APC process) CX-150A (APC process)
VTC-5300P series VTC-5400P series VTC-5000 series VTC-5150 series VTC-6500 series	LA7007 (Pre amp) LA7020 (REC process) LA7025 (PB process) LA7013 (Noise cancel)	M51437P (ACC process) M51438P (APC process) M50180P (APC-ID process) M51439P (PB SP process)
VTC-M10 series VTC-M20 series VTC-M30 series	LA7007S (Pre amp) LA7020 (REC process) LA7025 (PB process) LA7013 (Noise cancel)	M51436P (ACC process) M51438P (APC process) M50180P (APC-ID process) M51439P (PB SP process)
VTC-M40 series VTC-M50 series	LA7007S (Pre amp) LA7210 (Sync detector) LA7020 (REC process) LA7025 (PB process) LA7013 (Noise cancel)	TA7788AP (REC/PB process) TA7722P (PB SP process) TA7357AP (Sync separator)

Table 3-1

1. OPERATION DESCRIPTION OF VTC-M40
(1) PART NAMES AND FUNCTION






❹ Audio REC LEVEL controls
❺ Audio LEVEL meter
❸ AUDIO MODE indicator
Multiplex FILTER switch
❷ AUDIO OUT selector
❶ INPUT SELECTOR
❻ Audio REC MODE selector

❶ INPUT SELECTOR

Select the programme to be recorded.

TUNER: Video and audio from the built-in tuner section are recorded. (This selector should normally be set in this position.)

LINE: External video and audio input from the VIDEO and AUDIO IN terminals are recorded. (Channel indicator does not illuminate.)

SIMUL: For recording FM simulcast programmes.

❷ AUDIO OUT selector

Select an audio output mode during record or playback.

STEREO: Selects both the left and right channels of Beta hi-fi stereo sound. (This selector should normally be set in this position.)

LEFT: Selects the left channel of Beta hi-fi stereo sound.

RIGHT: Selects the right channel of Beta hi-fi stereo sound.

NORMAL: Selects the conventional monaural sound track.

❸ AUDIO MODE indicator

This indicator illuminates when recording in Beta hi-fi or when playing back a tape recorded in Beta hi-fi.

❹ Audio REC LEVEL controls

Adjust the audio recording levels (Left and Right) with these controls when the REC MODE selector is in the "MAN"position.

❺ Audio LEVEL meter

These meters show the peak input level of each channel (Left and Right) during Beta hi-fi recording, and recording levels during playback.

❻ Audio REC MODE selector

AUTO: Recording level is automatically adjusted to the optimum level. (This selector should normally be set in this position.)

MAN: Recording level can be adjusted manually by REC LEVEL controls (Left and Right).

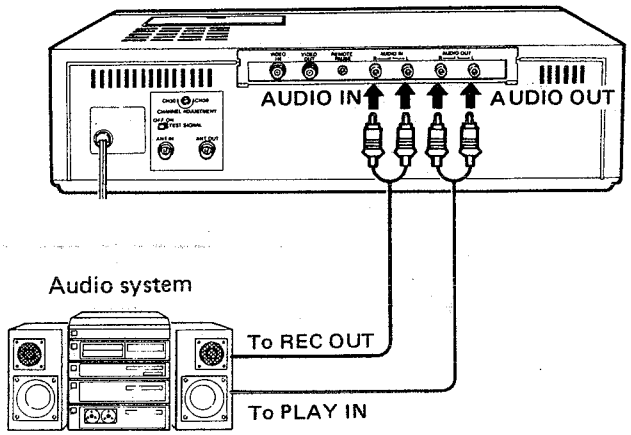
(2) BETA hi-fi AUDIO RECORDING

TO RECORD FROM THE AUDIO SYSTEM

You can record only audio signal without recording picture.

(1) **Turn the power on. (TIMER switch must be "OFF")**
(2) **Insert a video cassette.**
(3) **Set the INPUT SELECTOR to the "LINE" position.**
(4) **Set the AUDIO OUT selector to the "STEREO" position.**
(5) **Set the REC MODE selector to either the "AUTO" position or the "MAN" position.**
When the "MAN" position is selected, it is necessary to adjust the recording level.
(6) **Turn the power on and play the programme source to be recorded.**
(7) **Start recording.**

CONNECTIONS

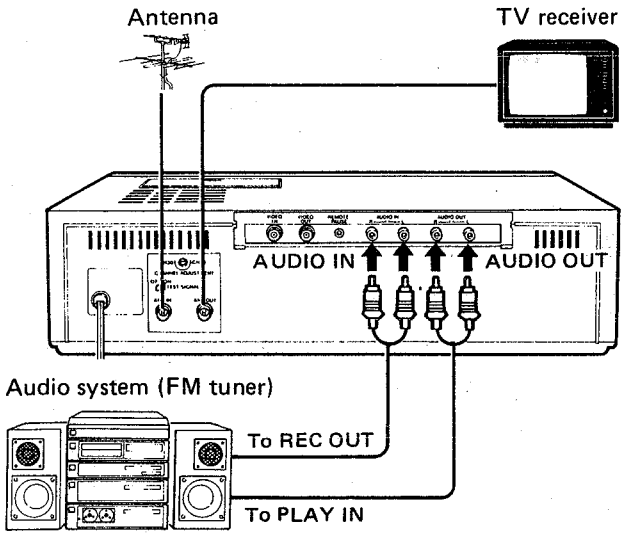


FM SIMULCAST RECORDING

You can record the FM simulcast TV programmes (stereo sound transmitted from an FM radio station) while recording the TV programme pictures (video signals) from the TV aerial inputs.

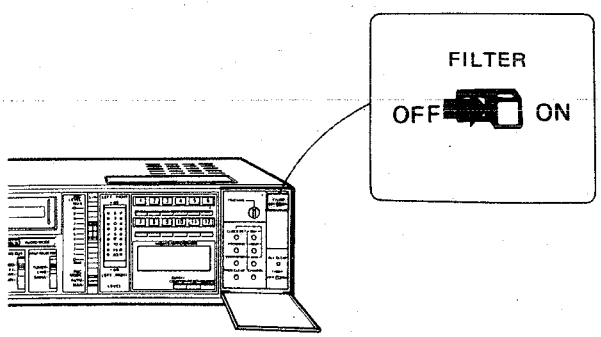
(1) **Turn the power on. (TIMER switch must be "OFF")**
(2) **Insert a video cassette.**
(3) **Set the INPUT SELECTOR to the "SIMUL" position.**
(4) **Set the AUDIO OUT selector to the "STEREO" position.**
(5) **Set the REC MODE selector to either the "AUTO" position or the "MAN" position.**
When the "MAN" position is selected, it is necessary to adjust the recording level.
(6) **Tune in the station you wish on the recorder and the FM tuner.**
(7) **Start recording.**

CONNECTIONS



Multiplex FILTER switch

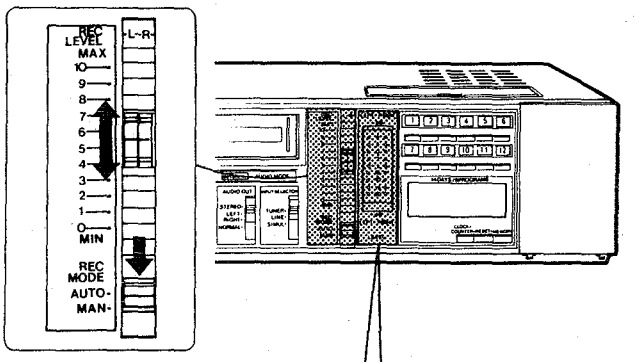
OFF: This switch should normally be set in this position.
ON: Set to this position if the sound is noisy when an FM tuner is used to record and playback an FM stereo broadcast.
Depending upon the FM tuner used, the 19-kHz pilot signal may leak during recording, thus weakening the effect of the noise-reduction system and resulting in noisy sound. If this occurs, setting to the "ON" position activates the MPX filter, thus cutting the 19-kHz pilot signal and making a better recording possible.



(3) RECORDING LEVEL ADJUSTMENTS

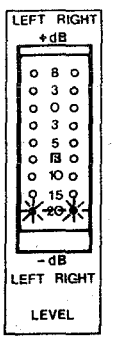
When recording on the Beta hi-fi system, the recording level of the left and right channels can be adjusted with REC LEVEL controls.

(1) Set the REC MODE selector to the "MAN" position.
(2) Adjust the left and right REC LEVEL controls.
The audio level meter should not illuminate beyond "0 dB" at the maximum input signal.

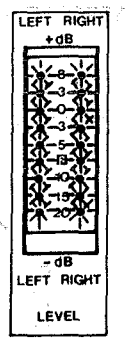


Readjust the REC LEVEL controls when the audio level meter are in the following conditions.

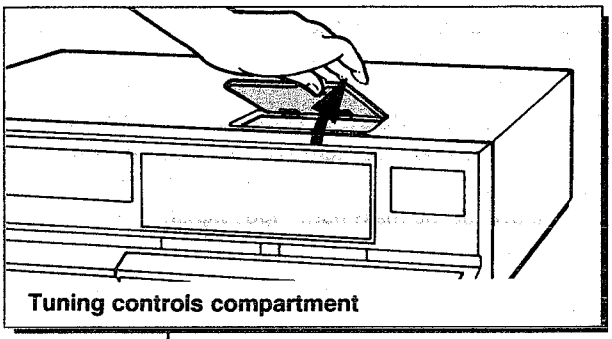
Settings are too low. . .
Recorded sound will be too low and unclear.

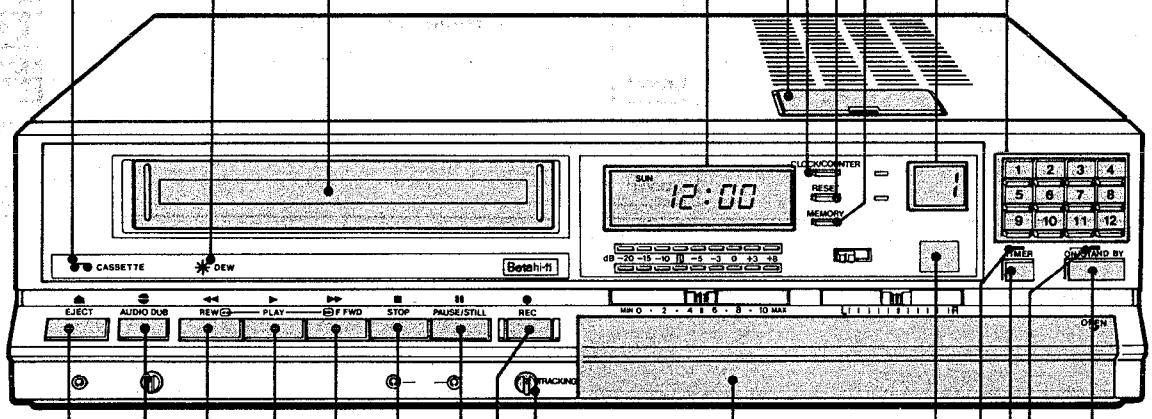


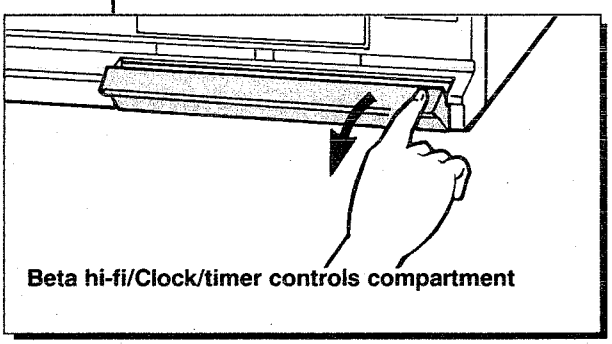
Settings are too high. . .
Recorded sound will be distorted.

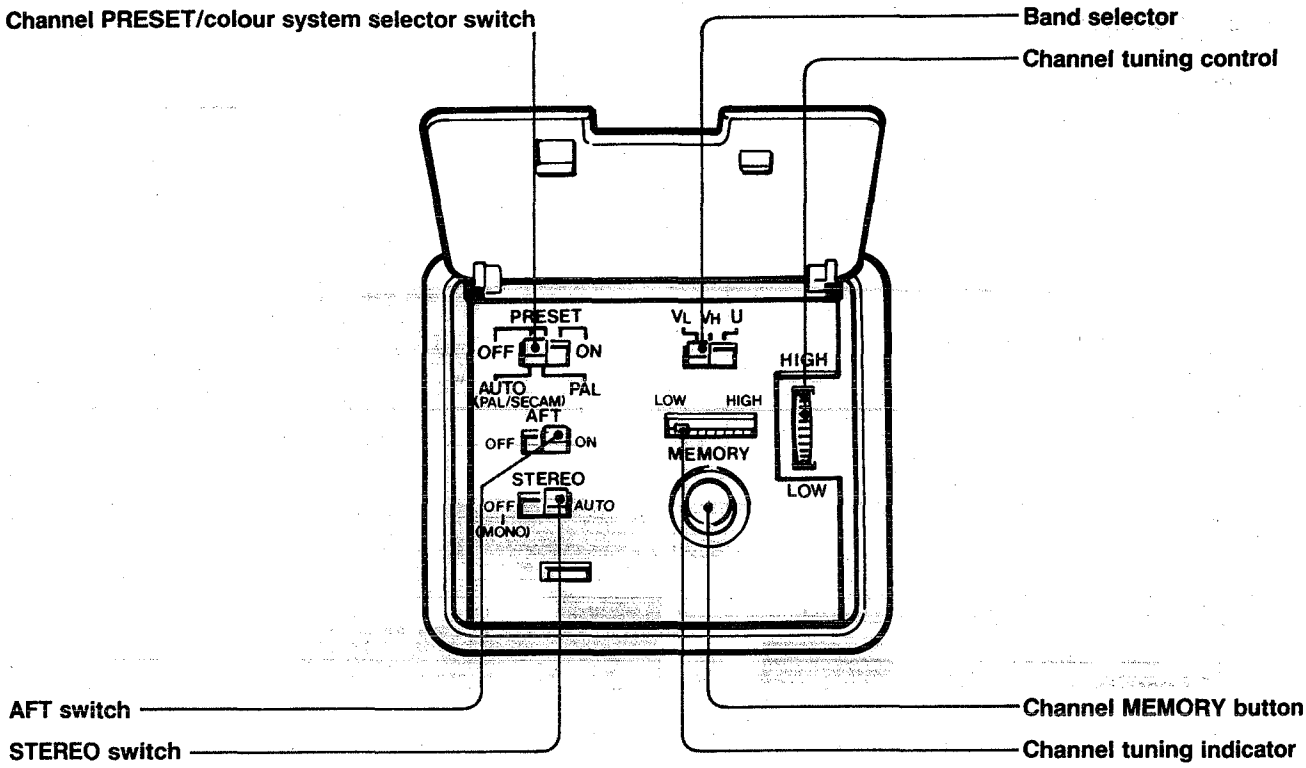


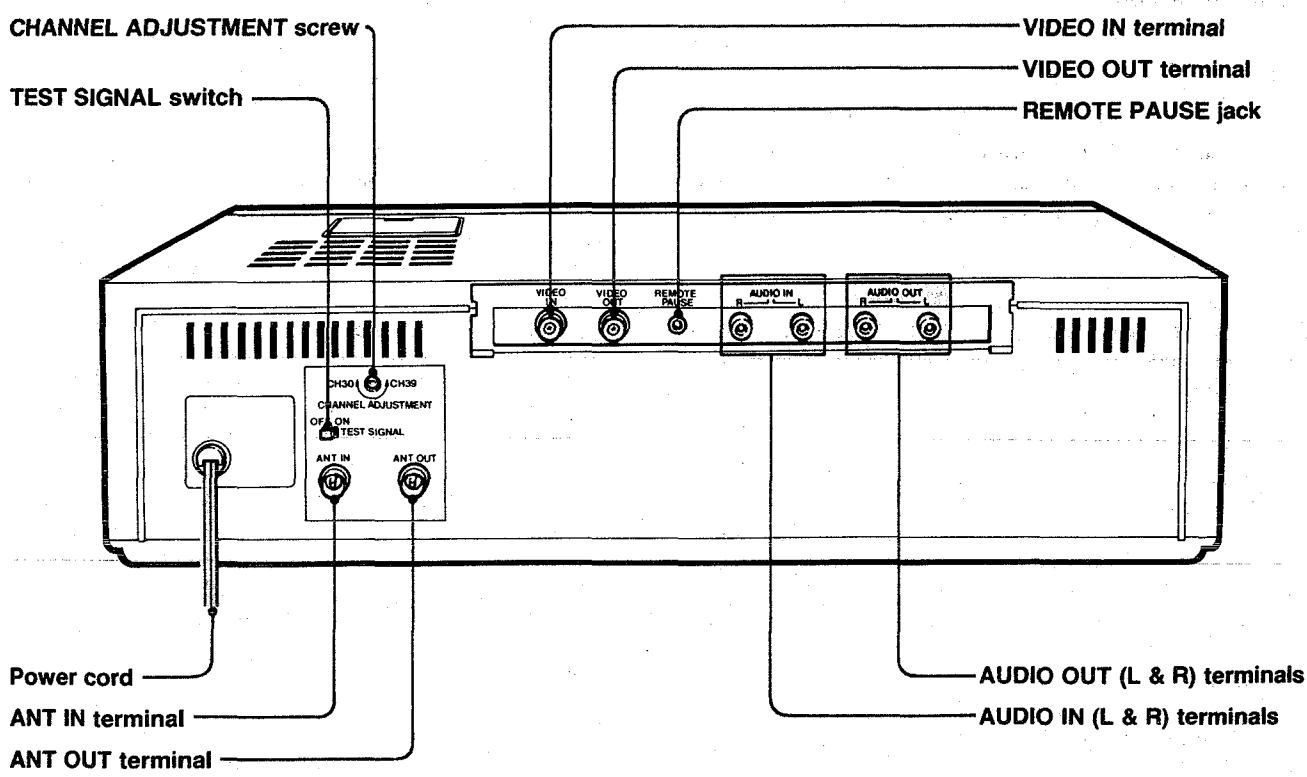
2. OPERATION DESCRIPTION OF VTC-M50
(1) PART NAMES AND FUNCTION



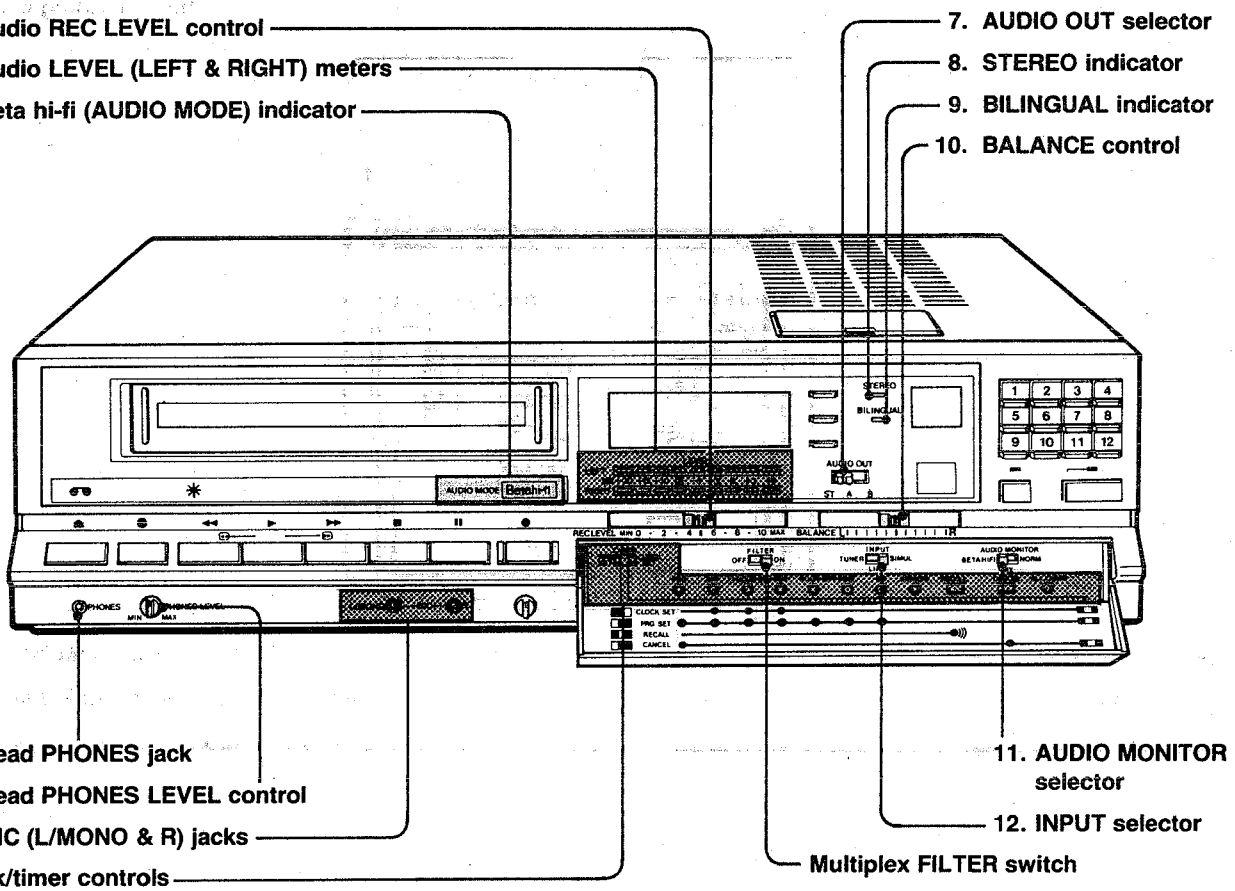





Channel PRESET/colour system selector switch
Band selector
Channel tuning control
PRESET
OFF
ON
AUTO (PAL/SECAM)
PAL
AFT
OFF
ON
STEREO
OFF
AUTO (MONO)
VL VH U
LOW
HIGH
MEMORY
HIGH
LOW
AFT switch
STEREO switch
Channel MEMORY button
Channel tuning indicator


CHANNEL ADJUSTMENT screw
TEST SIGNAL switch
VIDEO IN terminal
VIDEO OUT terminal
REMOTE PAUSE jack
VIDEO IN
VIDEO OUT
REMOTE PAUSE
AUDIO IN
AUDIO OUT
CH30
CH39
CHANNEL ADJUSTMENT
OFF ON
TEST SIGNAL
ANT IN
ANT OUT
Power cord
ANT IN terminal
ANT OUT terminal
AUDIO OUT (L & R) terminals
AUDIO IN (L & R) terminals

Beta hi-fi controls


1. Audio REC LEVEL control
2. Audio LEVEL (LEFT & RIGHT) meters
3. Beta hi-fi (AUDIO MODE) indicator
7. AUDIO OUT selector
8. STEREO indicator
9. BILINGUAL indicator
10. BALANCE control
4. Head PHONES jack
5. Head PHONES LEVEL control
6. MIC (L/MONO & R) jacks
Clock/timer controls
11. AUDIO MONITOR selector
12. INPUT selector
Multiplex FILTER switch

1. Audio REC LEVEL control

Adjust the audio recording levels (left and right) with this control. When set at the "▮" mark (click point), an appropriate sound level will be obtained for recording of normal broadcasts.

2. Audio LEVEL (LEFT & RIGHT) meters

These meters show the peak input level of each channel (left and right) during Beta hi-fi recording, and recording levels during playback.

3. Beta hi-fi (AUDIO MODE) indicator

This indicator illuminates when recording in Beta hi-fi or when playing back a tape recorded in Beta hi-fi.

4. Head PHONES jack (3.5mm dia. mini-jack)

When you use 8 ohms headphones (sold separately), connect them to this jack.

5. Head PHONES LEVEL control

This control is used to adjust the level of the sound you hear through the headphones.

6. MIC (L/MONO & R) jacks (3.5mm dia. mini-jack)

When using low impedance microphones (sold separately) to make a stereo recording, connect them to these jacks. If only one microphone is to be used, connect it to the MIC (L/MONO) jack. The recording will then be made monaurally.

7. AUDIO OUT selector

Select an audio output mode during record or playback.

ST: Selects both the left and right channels of Beta hi-fi stereo sound. (This selector should normally be set in this position.)

A: Selects the left channel of Beta hi-fi stereo sound. During a bilingual broadcast, selects the A (main) sound.

B: Selects the right channel of Beta hi-fi stereo sound. During a bilingual broadcast, selects the B (sub) sound.

Note: The AUDIO OUT selector will not function when the AUDIO MONITOR selector is set to the "NORM" position.

8. STEREO indicator
Illuminates when a stereo broadcast is being received.

9. BILINGUAL indicator
Illuminates when a bilingual broadcast is being received.

10. BALANCE control
Should usually be set to the centre position (click point). When adjusting the recording level manually during a Beta hi-fi recording, make the setting in accordance with the LEFT and RIGHT audio LEVEL meters.

11. AUDIO MONITOR selector
This selector can be used to select the hi-fi track sound or ordinary (non hi-fi) track sound when playing or recording.

BETA HIFI: Set to this position to playback only the hi-fi sound of a Beta hi-fi tape. During recording the sound is recorded on the hi-fi track and can be monitored.

MIX: Set to this position to playback a tape recorded in Beta hi-fi sound so that its hi-fi sound and ordinary sound are mixed. During recording the sound is recorded on the hi-fi track and can be monitored.

NORM: Set to this position to playback a tape recorded in Beta hi-fi sound as only ordinary sound. During recording the sound is recorded on the ordinary track and can be monitored.

12. INPUT selector
Select the programme to be recorded.

TUNER: Video and audio from the built-in tuner section are recorded. (This selector should normally be set in this position.)

LINE: External video and audio input from the VIDEO and AUDIO IN terminals are recorded. (Channel indicator does not illuminate.)

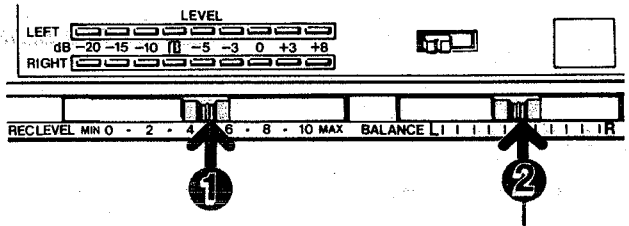
SIMUL: For recording FM simulcast programmes.

(2) AUDIO RECORDING LEVEL ADJUSTMENTS

When sound recording with the Beta hi-fi system, the audio REC LEVEL control should normally be set to the "▮" mark (click point) to obtain an appropriate sound level.
When recording from a microphone(s) or audio equipment, or any time there is significant difference between the highest and lowest volume levels of the material to be recorded, manual adjustment of the recording level is suggested for best results.

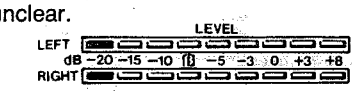
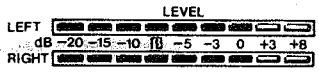
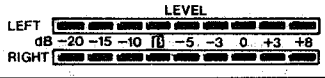
(1) Adjust the audio REC LEVEL control.
The audio LEVEL meters should not illuminate beyond "0 dB" at the maximum input signal.

(2) If there is a difference between left/right recording level, adjust the BALANCE control.
Adjust so that the illuminated length is the same for the LEFT and RIGHT parts of the audio LEVEL meters.



BALANCE CONTROL
* The BALANCE control functions only when recording mode.
* After use, be sure to return the BALANCE control to the centre position (click point).

Readjust the audio REC LEVEL control when the audio LEVEL meters are in the following conditions.

Setting is too low. . .	Setting is good...	Setting is too high...
Recorded sound will be too low and unclear.	Recorded sound will be best.	Recorded sound will be distorted.
		

(3) RECORDING STEREO/BILINGUAL BROADCASTS

For Beta hi-fi recordings, stereo/bilingual broadcasts can be recorded as they are. Stereo broadcasts will be recorded in stereo, and, for bilingual broadcasts, the main sound and sub sound will each be recorded. See "RECORDING TELEVISION BROADCAST" for basic information.

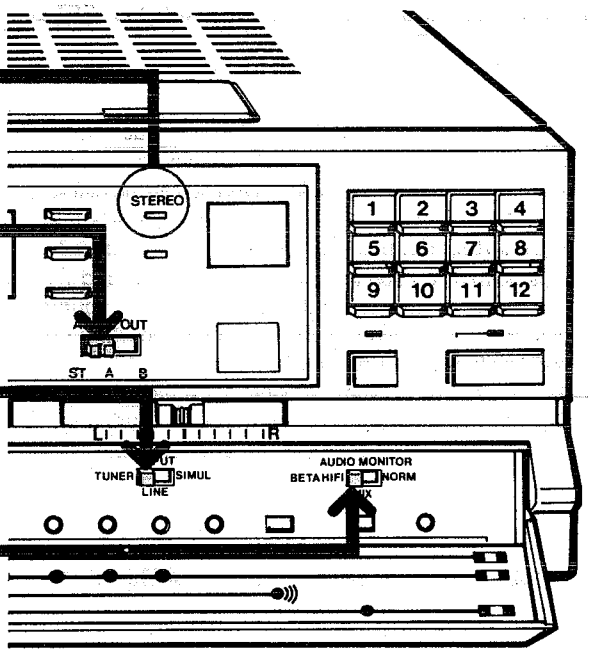
RECORDING STEREO BROADCASTS

STEREO indicator
* Illuminates when a stereo broadcast is being received.

AUDIO OUT selector
* Set to the "ST" position. Stereo broadcasts can be monitored.

INPUT selector
* Set to the "TUNER" position.

AUDIO MONITOR selector
* Set to the "BETA HI-FI" position.



Left and right sounds are recorded mixed on the normal audio track.

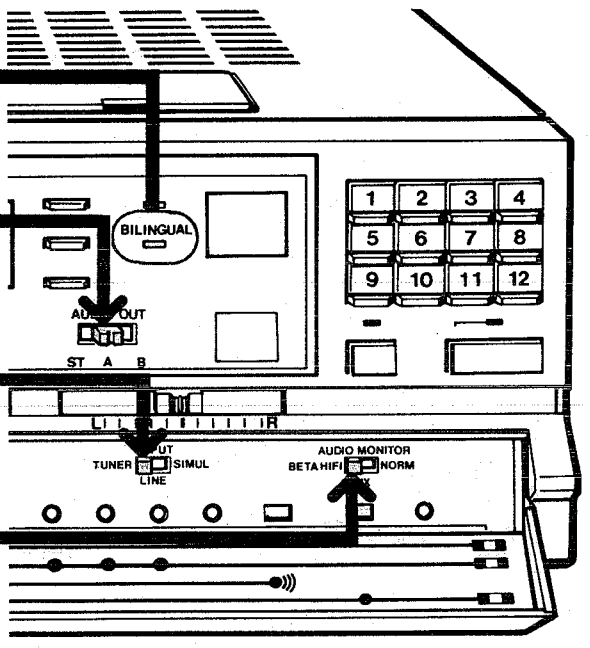
RECORDING BILINGUAL BROADCASTS

BILINGUAL indicator
* Illuminates when a bilingual broadcast is being received.

AUDIO OUT selector
* To monitor the A (main) broadcast, set to the "A" position.
* To monitor the B (sub) broadcast, set to the "B" position.

INPUT selector
* Set to the "TUNER" position

AUDIO MONITOR selector
* Set to the "BETA HI-FI" position.



The A (main) broadcast sound is recorded on the normal audio track.

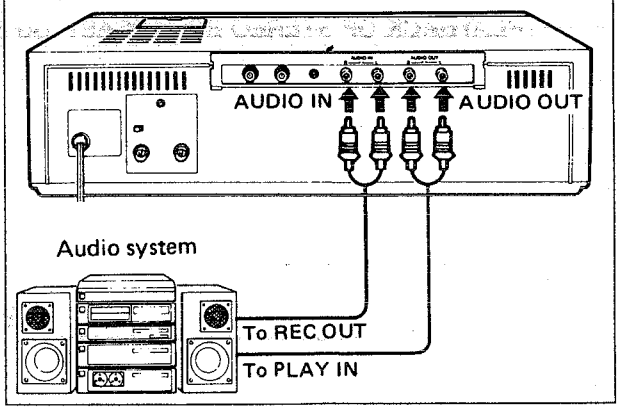
(4) BETA hi-fi AUDIO RECORDING

TO RECORD FROM THE AUDIO SYSTEM

You can record only audio signal without recording picture.

(1) Turn the power on.
(2) Insert a video cassette.
(3) Set the INPUT selector to the "LINE" position.
(4) Set the AUDIO OUT selector to the "ST" position.
(5) Turn the power on and play the programme source to be recorded.
(6) Start recording.

CONNECTIONS

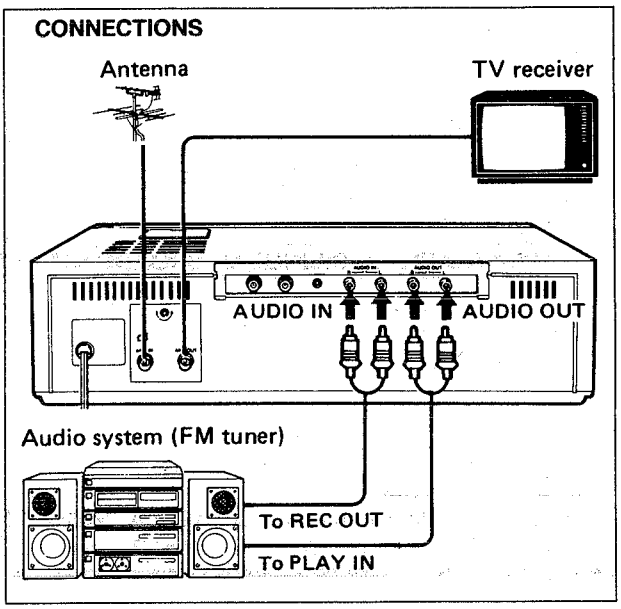


FM SIMULCAST RECORDING

You can record the FM simulcast TV programmes (stereo sound transmitted from an FM radio station) while recording the TV programme pictures (video signals) from the TV aerial inputs.

(1) Turn the power on.
(2) Insert a video cassette.
(3) Set the INPUT selector to the "SIMUL" position.
(4) Set the AUDIO OUT selector to the "ST" position.
(5) Tune in the station you wish on the recorder and the FM tuner.
(6) Start recording.

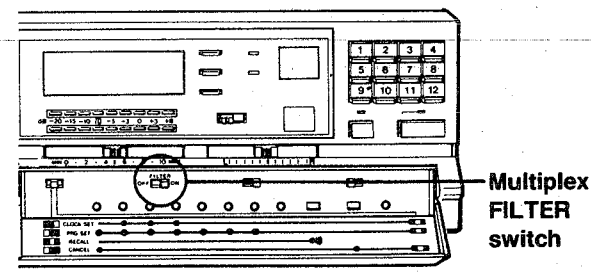
CONNECTIONS



When a video recording is made with the INPUT selector set to the "SIMUL" position, the sound recorded on the normal audio track will be, regardless of line input, a mixture of the left and right sounds of the broadcast received by the VCR's tuner (stereo broadcast) or the A (main) broadcast sound (bilingual broadcast).

Multiplex FILTER switch

OFF: This switch should normally be set in this position.
ON: Set to this position if the sound is noisy when an FM tuner is used to record and playback an FM stereo broadcast. Depending upon the FM tuner used, the 19-kHz pilot signal may leak during recording, thus weakening the effect of the noise-reduction system and resulting in noisy sound. If this occurs, setting to the "ON" position activates the multiplex filter, thus cutting the 19-kHz pilot signal and making a better recording possible.



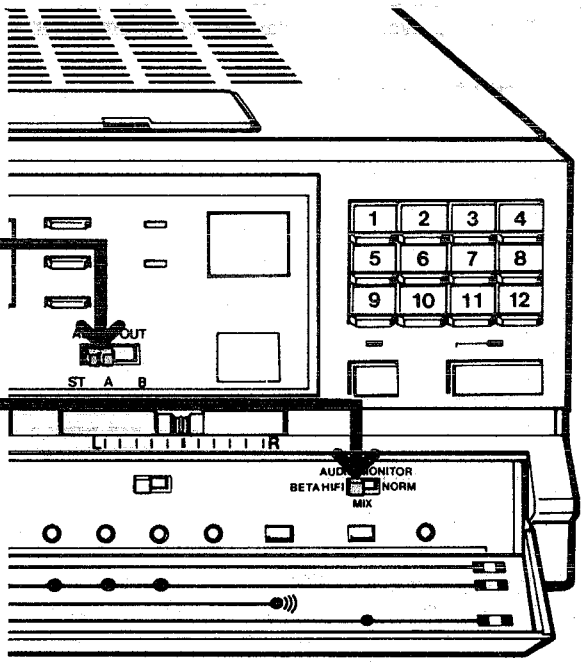
(5) PLAYBACK STEREO/BILINGUAL BROADCAST TAPE

Basic operations are the same as described in "NORMAL PLAYBACK".
You can enjoy powerful playback sound by using a combination of a dual TV, stereo AV colour TV or regular colour TV, plus stereo equipment and headphones.

PLAYBACK OF STEREO BROADCAST TAPES—Playback of tapes recorded in Beta Hi-Fi

AUDIO OUT selector
* Set to the "ST" position.

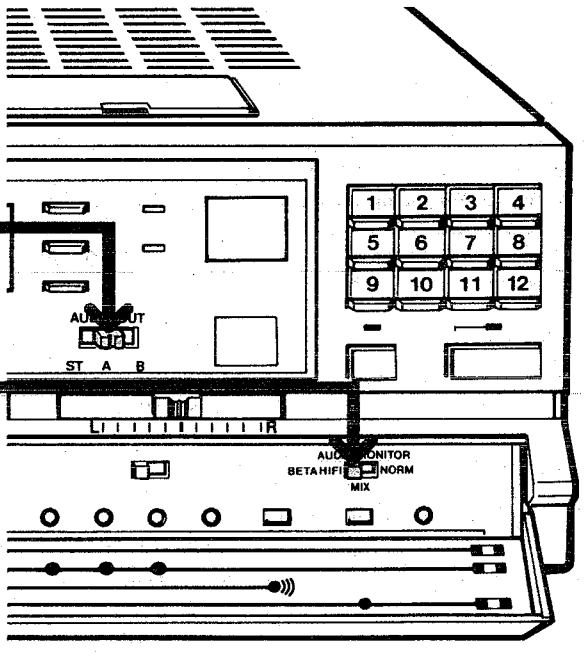
AUDIO MONITOR selector
* Set to the "BETA HIFI" position.



PLAYBACK OF BILINGUAL BROADCAST TAPES—Playback of tapes recorded in Beta Hi-Fi

AUDIO OUT selector
* To monitor the A (main) broadcast, set to the "A" position.
* To monitor the B (sub) broadcast, set to the "B" position.

AUDIO MONITOR selector
* Set to the "BETA HIFI" position.

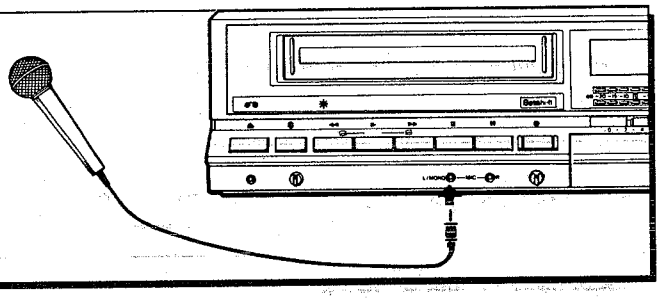


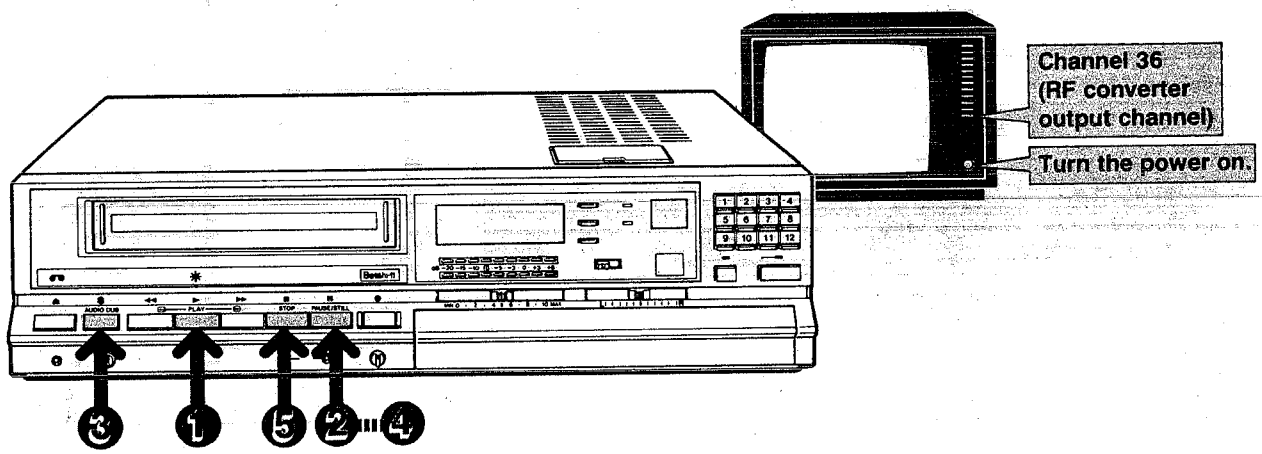
(6) AUDIO DUBBING

New sounds (such as commentary, music or special effects) can be added to the sound track of an already recorded tape. Note that existing sounds for that part of the tape will be erased when the new sounds are added.
Beta hi-fi recordings cannot be made. The sound is added to the ordinary sound track only, as monaural sound.

Preparations

* If a microphone is to be used, connect it to the MIC (L/MONO) jack.
* If audio equipment is to be used, connect it to the AUDIO IN (L) and (R) terminals. (Set the INPUT selector to the "LINE" position.
* If a microphone and audio equipment are connected at the same time, only the sound from the microphone will be recorded. (The microphone has priority.)





(1) **Playback the tape.**
Playback the tape on which the audio dubbing is to be made and find the place where it is to start.

AT THE PLACE WHERE DUBBING IS TO BEGIN

(2) **Press the PAUSE/STILL button.**
A still picture will be seen on the screen.

(3) **Press the AUDIO DUB button.**
The VCR will then be in the audio dubbing stand-by mode. (The still picture can still be seen on the screen.)
Select which sound you want to hear by referring to the table.

(4) **Press the PAUSE/STILL button once again.**
The audio dubbing will begin.
To temporarily stop an audio dub, press the PAUSE/STILL button.

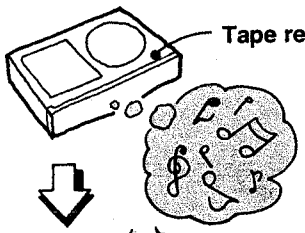
(5) **Press the STOP button to stop an audio dubbing.**
When the end of the tape is reached audio dubbing will stop and the tape automatically rewinds.

Notes:

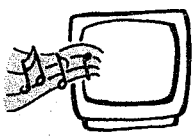
* If two microphones are used at the same time for audio dubbing, connect the second one to the MIC (R) jack. (Note that the sound recorded through the two microphones will be mixed together on the tape. This method makes recordings such as a duet possible.)
* The audio dubbing method cannot be used for recording of TV sound.

INTERESTING USES

* If a tape recorded in Beta hi-fi is used, the audio dub can be used as your practice of a foreign language conversation (model sentence recorded on video track, for example, or to add a sing-along accompaniment to music recorded on the video track).



plus



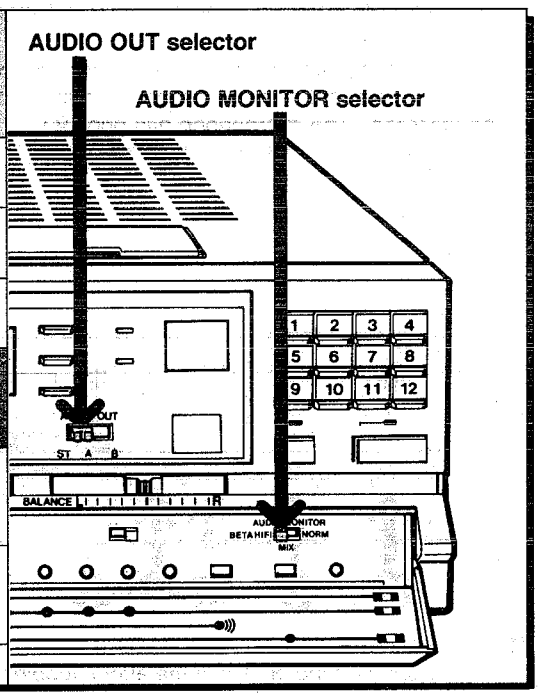
Sing-along

The sounds on both tracks (the original sounds on the video track and the new sounds on the ordinary sound track) can be heard mixed together by setting the AUDIO MONITOR selector to the "MIX" position. (To listen only to audio dubbed sound, set to the "NORM" position.)

TO MONITOR AUDIO DUBS IN PROGRESS OR LISTEN TO COMPLETED RESULTS

* For tapes which have the ordinary sound track (i.e., not Beta hi-fi), the audio dub can be monitored while it's being made, or played back for listening after the dub is finished, regardless of the position to which the AUDIO OUT selector and the AUDIO MONITOR selector are set.
* The previously recorded sounds on that part of the tape will have been erased and only the newly recorded sounds will remain.
* For audio dubs made on a tape on which the original recording was a Beta hi-fi recording however, use the AUDIO OUT selector and the AUDIO MONITOR selector to select which sound track you want to hear.
* The original sounds (Beta hi-fi sounds) on the video track will not have been disturbed by the audio dub; only the sounds on the ordinary sound track will have been re-recorded.

Sound to be heard	AUDIO MONITOR selector setting	AUDIO OUT selector setting
Video track "L and R" or "A (main) and B (sub)" sound	BETA HIFI	ST
Video track "L" or "A (main)" sound	BETA HIFI	A
Video track "R" or "B (sub)" sound	BETA HIFI	B
Audio dubbed sound and video track "L and R" or "A (main) and B (sub)" sound mixed	MIX	ST
Audio dubbed sound and video track "L" or "A (main)" sound mixed	MIX	A
Audio dubbed sound and video track "R" or "B (sub)" sound mixed	MIX	B
Audio dubbed sound only	NORM	Any position



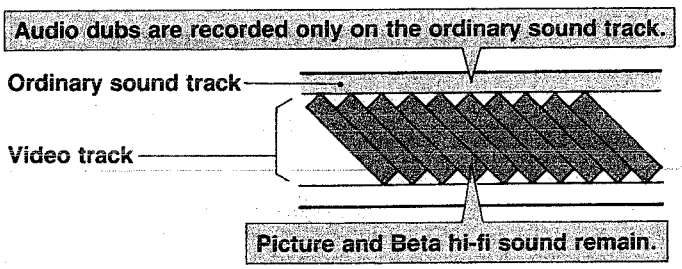
* After the audio dubbing is finished, return the AUDIO MONITOR selector to the "BETA HIFI" (or "NORM") position.

Notes:

* The mode changes to the audio-dubbing mode if the AUDIO DUB Button and the PLAY Button are pressed during the STOP mode condition.
* Audio dubs cannot be made if the erasure-prevention tab on the cassette has been broken out.
* When an audio dub such as described above is made, only the sound is recorded, so the AUDIO OUT selector can be set to any position.

HOW AUDIO DUBS ARE MADE

* For audio dubs, the new sound recording is made only on the ordinary (monaural) sound track. When, therefore, an audio dub is added to a tape originally recorded in Beta hi-fi, the picture and sound (Beta hi-fi sound) on the video track are not erased.


Audio dubs are recorded only on the ordinary sound track.
Ordinary sound track
Video track
Picture and Beta hi-fi sound remain.

2. AUDIO CIRCUIT

1. AUDIO SIGNAL SYSTEM CIRCUITRY

The audio signals of ordinary, home-type VCRs, in the same way as audio tape recorders, is recorded by employing AC bias, using the fixed-head system. In the PAL Beta system, as shown in Fig. 2-1-1, the audio signals are recorde on the audio track at the upper edge of the tape, employing AC bias by the fixed-head system, in the same way as for ordinary audio tape recorders.

However, because the audio system of home-type VCRs has certain features and characteristics peculiar only to VCRs, this system is explained below.

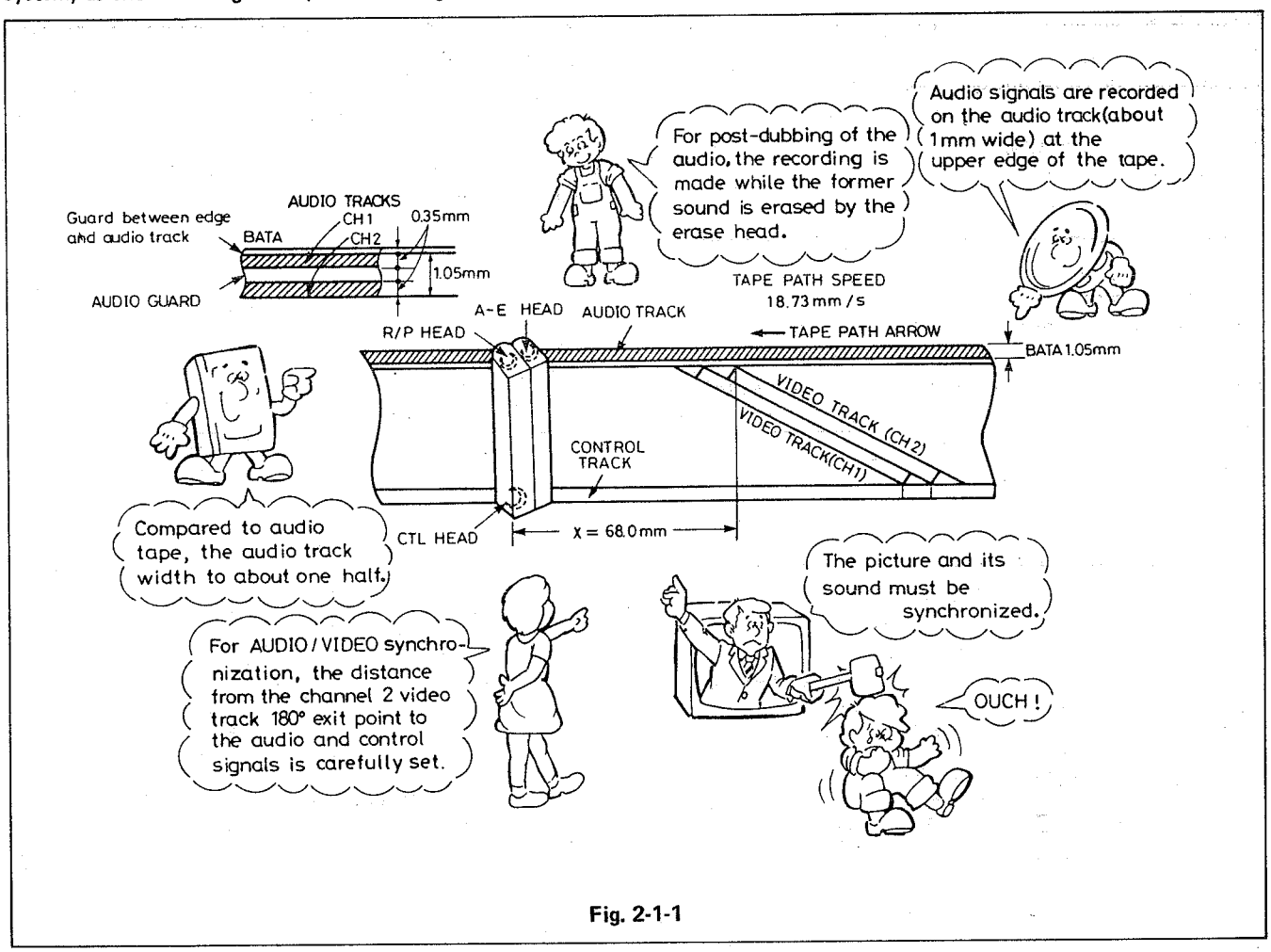

Audio signals are recorded on the audio track(about 1mm wide) at the upper edge of the tape.
For post-dubbing of the audio, the recording is made while the former sound is erased by the erase head.
AUDIO TRACKS
CH1
CH2
0.35mm
1.05mm
Guard between edge and audio track
BATA
AUDIO GUARD
TAPE PATH SPEED
18.73mm/s
A-E HEAD
AUDIO TRACK
R/P HEAD
TAPE PATH ARROW
BATA 1.05mm
VIDEO TRACK (CH2)
VIDEO TRACK(CH1)
CONTROL TRACK
CTL HEAD
X = 68.0mm
Compared to audio tape, the audio track width to about one half.
The picture and its sound must be synchronized.
For AUDIO/VIDEO synchronization, the distance from the channel 2 video track 180° exit point to the audio and control signals is carefully set.
OUCH!

Fig. 2-1-1

(1) TV PICTURE AND SOUND ARE MADE TO AGREE

If the interval on the tape between the video signals and the audio signals recorded on the VCR were not perfectly uniform for various models and brands of VCRs, recorded tapes would not be compatible for interchangeable use from one VCR to another, because there would be an annoying deviation between the video movements seen on the screen and the sounds which accompany those movements.

The agreement of timing of these audio signals and video signals in known as "ripple sync", and, considering the installation position of the audio head for the tape format shown in Fig. 2-1-1 as X, its distance from the video heads is strictly prescribed by standards.

(2) AUDIO DUBBING (AFTER RECORDING) OF AUDIO SIGNALS IS POSSIBLE.

"Audio dubbling" is the adding of new audio signals to the tape on which video and audio signal have already once been recorded, those new audio signals thus replacing the former audio signals.

For such dubbing, the video system is operated in the playback mode while the audio system is operated in the recording mode, with the audio recording being made on the audio track immediately after the former audio signals have been erased by the erase head (located just prior to the audio recording head).

For ordinary recording on the other hand, both the full--erase head and the audio erase head function. When video signals are recorded by the VCR, the adjustment of the video recording level is automatically performed by the AGC circuitry. For the audio signals as well, the ALC circuitry (Automatic Level Control; functions in the same way as the AGC circuitry) is used for automatic adjustment of the level of audio signals being recorded.

(3) MUTING

Muting of the audio signals is an operation used in order to prevent playback of imperfect sounds which occur when, during the loading of the tape onto the rotating head drum, the audio head comes into contact with the tape. As a result, audio signals are suppressed during the loading of the tape onto the rotating head drum, and, after loading is completed, signals are immediately output.

In the same way, the output of audio signals is also interrupted when the pause button is pressed and until the pause mode is released.

(4) NOISE REDUCTION

As can be seen in Fig. 2-1-1, the width of the audio track is approximately 1.05 mm for monaural use. The tape speed is 18.73 mm/second. If we compare this to the width of tape used in an audio cassette recorder, we find that the stereo track width in the audio tape is 0.6 mm and the tape speed is 47.6 mm/second, which means that the track width of the audio track of a VCR is narrower and the tape speed is slower, so that the conditions are therefore in fact quite severe for tone quality.

In other words, the signal-to-noise ration falls as the track width becomes narrower, and the frequency response falls as the tape speed becomes slower.

In order to overcome these problems then, the audio head used in VCRs has a narrow gap for high performance characteristics, and, moreover, noise-reduction circuitry is employed in order to improve tone quality.

The Dolby noise-reduction system employed in audio cassette recorders is of course very well known, the fundamental principle of the operation of this type of circuit being to make the recording while emphasizing the high range of the low levels in which noise is a problem, and then, during playback, to conversely return the signal to the original condition and in this way attempt to suppress noise.

Figure 2-1-2 shown the encoding characteristics (the relationship between the output corresponding to the level and the input frequency at the recording side) and the decoding characteristics (the relationship between the output corresponding to the level and the input frequency at the playback side) of the noise-reduction circuitry.

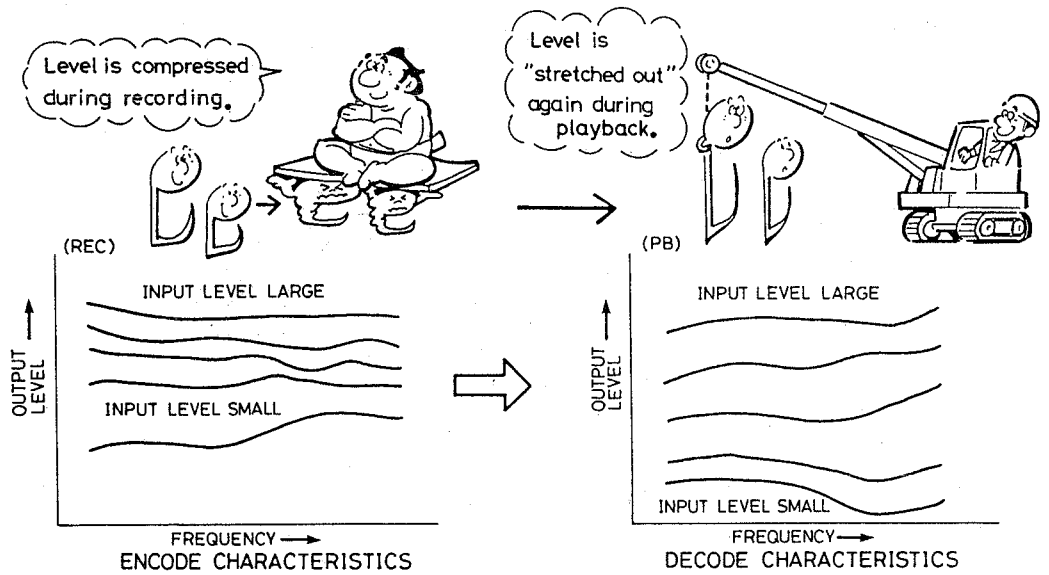


Fig. 2-1-2

(5) AUDIO CIRCUITRY CONFIGURATION

Figure 2-1-3 shows an example of the configuration of actual audio circuitry.

If we trace the path of signal flow in this circuitry, the audio signal (during recording) passes from the input through the head equalizer amplifier, and, after automatic adjustment of the level at the ALC circuit, the signal is amplified while compensation is made for the high range and low range at the equalization amplifier. The output signals of the equalization amplifier pass through the bias trap, are mixed with 70-kHz AC bias from the bias oscillator, and are supplied to the audio head. The output from the bias oscillator is simultaneously supplied to the full-erase head and to the audio erase head, where signal erasure is performed.

During playback, playback signals from the audio head are low-range compensated at the playback equalization circuit, and, after amplification, pass through the muting circuit and are output.

During playback, the bias-oscillation circuit and the ALC circuit do not function.

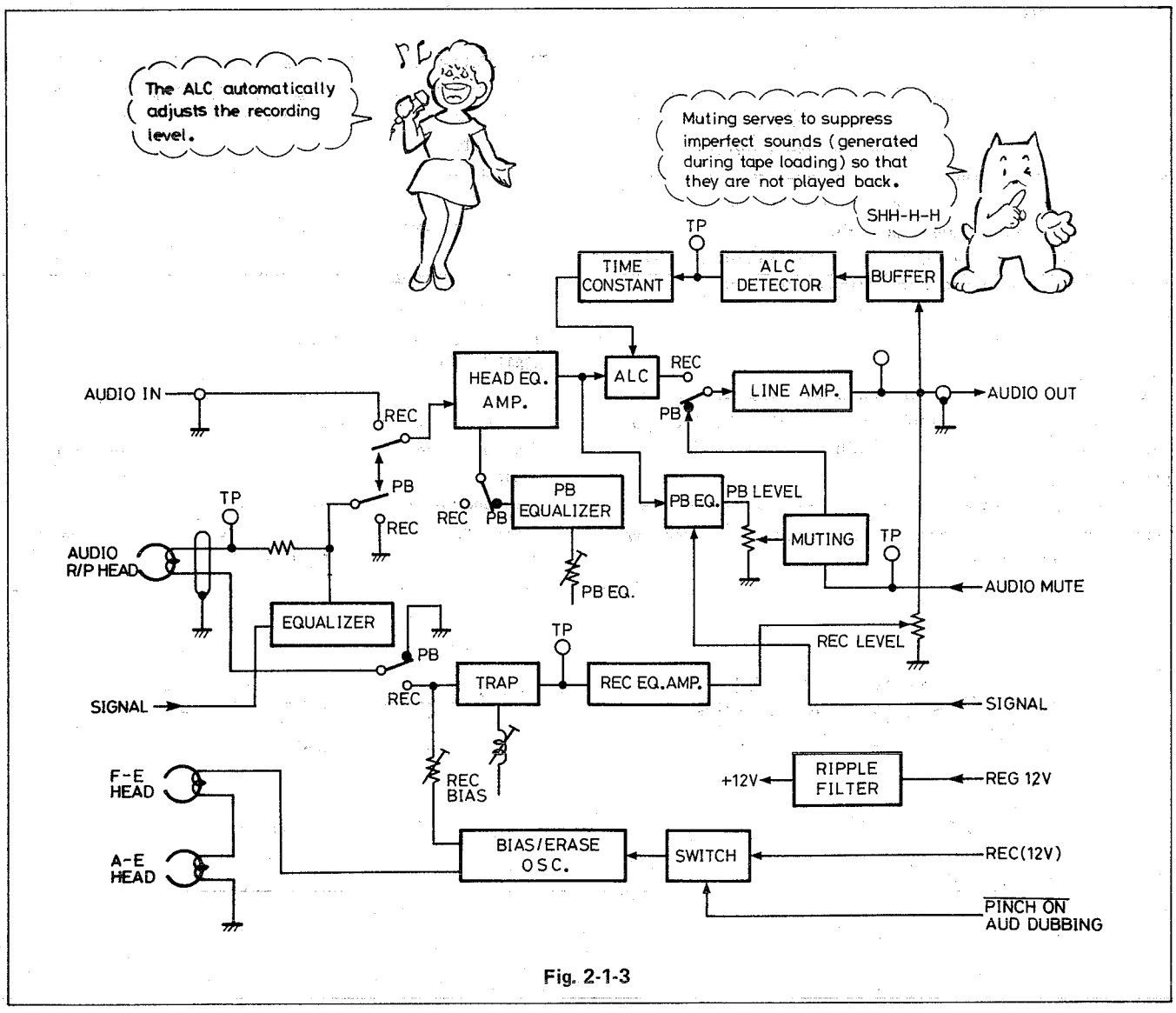


Fig. 2-1-3

2. AUDIO hi-fi VIDEO

"hi-fi" means of course high fidelity, which means that original sound (the sound source) is faithfully recorded and reproduced. "hi-fi video" therefore refers to video (or a VCR) which is characterized by especially high tone quality.

Among "hi-fi videos" there are the Beta system and the VHS system, respectively call "Beta hi-fi" and "VHS hi-fi".

For the PAL system, fundamentally the same method-deep recording-is employed for both Beta hi-fi and VHS hi-fi.

For hi-fi video, compared to conventional video, the following points are established regarding the audio circuitry names is order to distinguish it from the hi-fi circuitry.

- The conventional type of audio circuitry is called the "normal audio circuitry", and the audio track on the tape are called "linear audio track" or "normal audio track".
- The Beta hi-fi audio circuitry is called the "audio hi-fi circuitry" and the audio tracks on the tape are called the "audio hi-fi tracks" or the "audio FM tracks".

There are then, in order to maintain the compatible coexistence of hi-fi video and conventional video, the "normal audio circuitry" and the "audio hi-fi circuitry", and the audio hi-fi circuitry serves also to handle stereo broadcasts and bilingual broadcasts.

There is, moreover, an input selection circuit which functions to make possible even more convenient functions of the hi-fi VCRs which use this circuitry.

(1) AUDIO hi-fi

Ordinary audio signals in a VCR are recorded, in the same way as an audio tape recorder, by employing AC bias by the fixed-head system.

For this tape of system there is an inevitable deterioration of frequency response, wow & flutter, etc. as a result of the reduction of tape speed used in order to obtain longer recording/playback time from the tape, thus making it difficult to obtain high tone quality. The tape speed employed for the PAL Beta system is 18.73 mm/second, which is, compared to the tape speed (47.6 mm/second) of audio tape recorders, only 1/4 to 1/2 of that speed.

It is for this reason that the rotating head system is utilized for the recording and playback of video signals, and to this a head exclusively for audio is added, resulting in an audio hi-fi system which makes possible the recording and playback of audio signals.

The relative speed of the tape versus the rotating head is 5.832 meters per second, giving a very high speed which is approximately 300 times that obtained by using a fixed-head system.

This can be imagined more clearly perhaps if we compare the speed of a man walking (the tape speed by using the fixed-head system) with the speed of a jet airplane (the relative tape speed by using the rotating head system). The result is, as shown in Table 2-2-1, that the tone quality of hi-fi video is comparable with that of the compact disc.

PERFORMANCE COMPARISON: hi-fi VCR AND COMPACT DISCS

	Audio hi-fi VCR	Compact disc
Frequency response	20 ~ 20,000 Hz	4 ~ 20,000 Hz
Dynamic range	80dB or higher	90dB or higher
Distortion factor	0.3% or less	0.004 or less
Wow and flutter	0.005% or less	0
Separation	60dB or higher	90dB or higher

Table 2-2-1

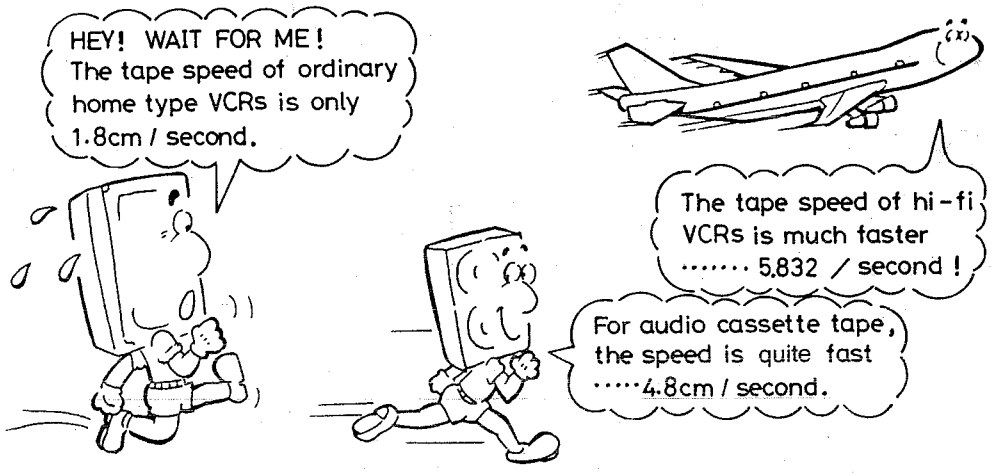

HEY! WAIT FOR ME!
The tape speed of ordinary home type VCRs is only 1.8cm / second.
The tape speed of hi-fi VCRs is much faster ····· 5.832 / second !
For audio cassette tape, the speed is quite fast ·····4.8cm / second.

Fig. 2-2-1

(2) OUTLINE OF THE RECORDING SYSTEM

The maximum frequency which can be recorded on magnetic tape is proportional to the relative speed of the head and the magnetic tape, and, moreover, the narrow head gap makes higher recording frequencies possible.

Currently, head gaps on VCR heads can be precisely processed to a width as narrow as about 0.4μm. Consequently, the maximum recording frequency is determined by the relative tape-to-head speed. This relative tape-to-head speed is virtually determined by the diameter of the head drum and by the head's rotational speed; because the head's rotational speed is prescribed to be 50 Hz, the determination is therefore made by the diameter of the drum. Consequently, for the PAL Beta system, this speed is 5.832 meters/second.

In addition, the luminance signal band is wide for the PAL system, and, when this is FM modulated, sideband waves also become wide.

In other words, a wide frequency band is necessary for tape recording by the PAL system.

It is for this reason then that the video signals, which are high frequency, are only recorded to a shallow depth of the tape, and lower frequencies are recorded to a deeper depth, thereby taking advantage of the characteristics of magnetic recording and modulating audio signals at a lower frequency than video signals, thus resulting in the "deep recording" system in which the recording is made at the deeper part of the tape.

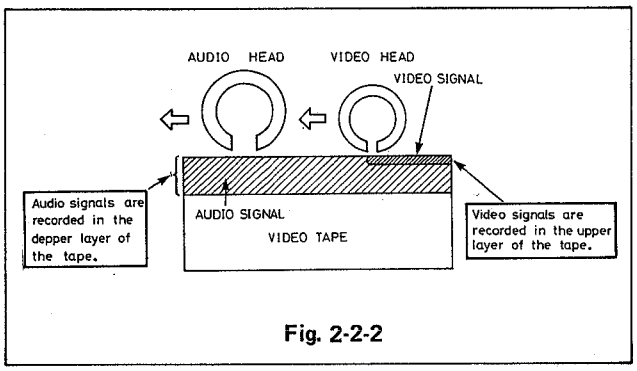

AUDIO HEAD
VIDEO HEAD
VIDEO SIGNAL
Audio signals are recorded in the depper layer of the tape.
AUDIO SIGNAL
VIDEO TAPE
Video signals are recorded in the upper layer of the tape.

Fig. 2-2-2

(3) PRINCIPLES OF DEEP RECORDING

For the audio hi-fi system, as shown in Fig. 2-2-3, there are, in addition to the video heads, two audio heads on the rotating disc, and audio signals are FM modulated and then recorded on the tape.

As shown in the figure, the audio signals precede the video signals, and are recorded deeper into the tape thickness.

Because the carrier wave for the audio signals is established as a lower frequency compared to the carrier wave for the luminance signals, the gaps of the audio heads are wider than the gaps of the video heads, with the result that magnetization is possible to a deeper part of the tape's magnetic layer.

The video signals are recorded on top of these audio signals, but, because the magnetic force in the deeper part of the tape becomes smaller as video signals are recorded on the surface of the tape when these video signals are recorded on top of already recorded audio signals, the audio signals remain, with the result that the two types of signals coexist together on the tape. Because the audio signals in the outer (shallow) layer of the tape are erased when audio signals are played back, there is demagnetization of the playback output by about 12 dB, but, because FM modulation is used, there is no problem regarding the effects of noise, etc.

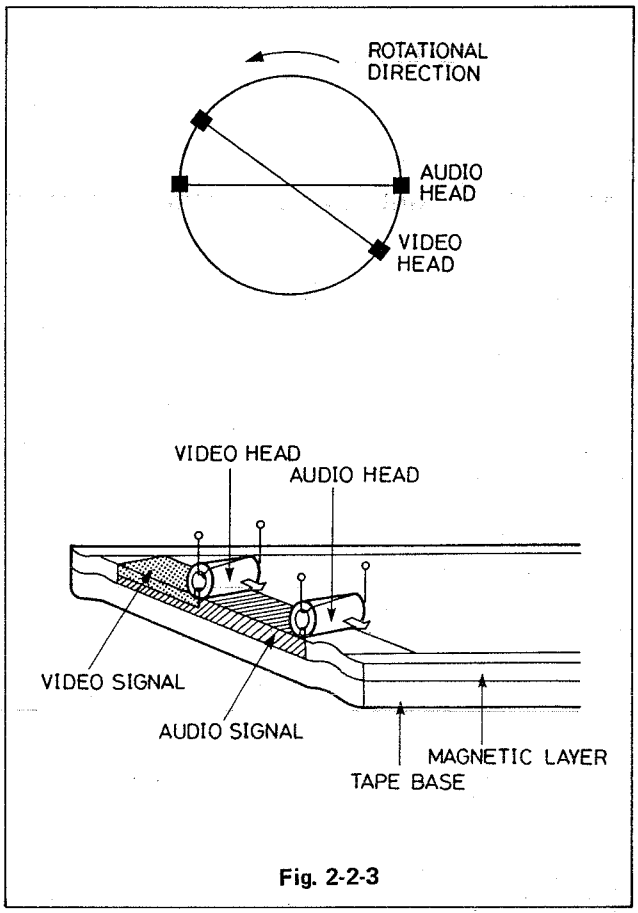


Fig. 2-2-3

Figure 2-2-4 shows, by a cross-section of the tape, the condition of recorded video signals. As shown in this figure, the tape usually consists of a coating of a 4 to 5 μm magnetic substance applied to a polyester base film, giving an overall tape thickness of about 19 μm.

Of that depth, the video signals recorded onto the tape by the video heads occupy only a small part of the surface of the magnetic material.

Figure 2-2-5 shows the relationship between the head gap (g) and the magnetic force (Hx) at a point which is distance (y) in the vertical direction from the maximum magnetic field (Ho) of the video heads.

This magnetic field inverts the larger part of the magnetic material which is larger than the anti-magnetic force (Hc) of the magnetic material, and the recording is made.

The anti-magnetic force (Hc) of ordinary video tape is Hc $\fallingdotseq$ 600 Oe (Oersted), and the magnetic field force (Ho) within the gap is Ho $\fallingdotseq$ 6000 Oe, with the result that Hx/Ho = 0.1. Consequently, the magnetic field distance by the head gap (g) is, according to the figure, y/g = 3.3.

If the gap of the video head is 0.4μm, y = 0.4 x 3.3 = 1.3μm, thus there will be no inversion of the magnetic material at a depth of 1.3 μm or more, and video signals are therefore not recorded at that depth. As a result, audio information can be recorded in the deeper part of the magnetic material where the video signals are not recorded.

CROSS-SECTION OF TAPE IN WHICH VIDEO SIGNALS ARE RECORDED

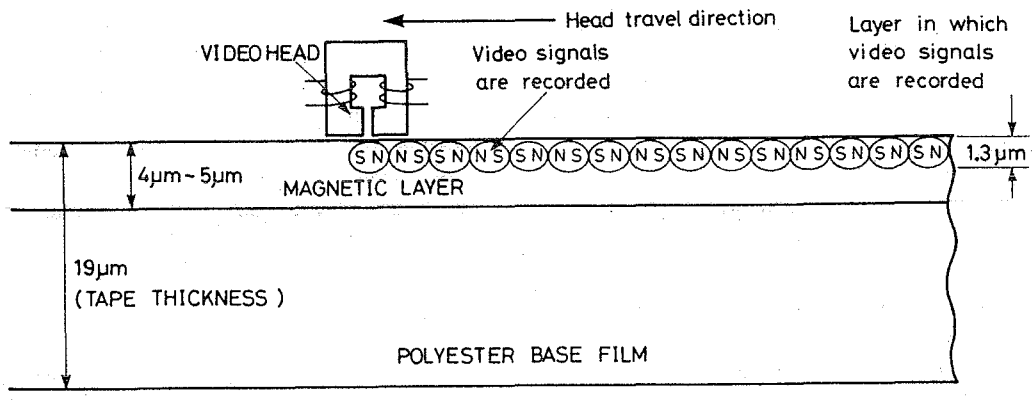

Head travel direction
VIDEO HEAD
Video signals are recorded
Layer in which video signals are recorded
S N N S S N N S S N N S S N N S S N N S S N N S S N
1.3μm
4μm~5μm
MAGNETIC LAYER
19μm
(TAPE THICKNESS)
POLYESTER BASE FILM

Fig. 2-2-4

RECORDING MAGNETIZATION DISTRIBUTION OF HEAD

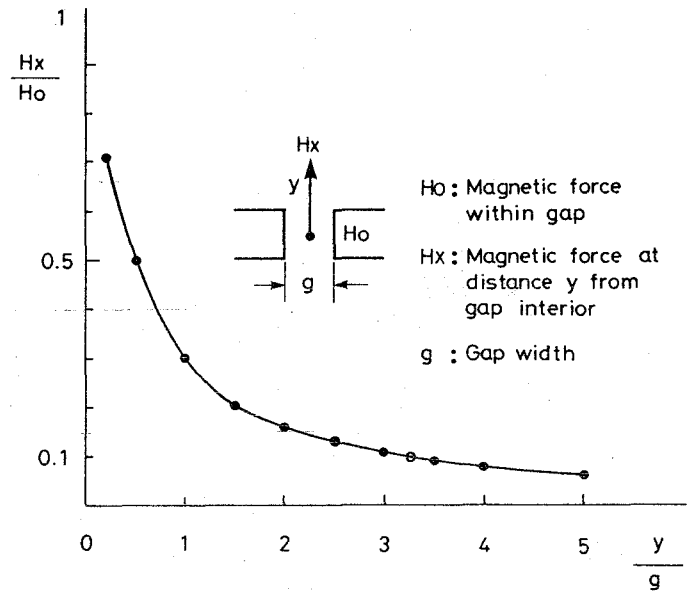


Fig. 2-2-5

As shown in Fig. 2-2-6, there are, for audio hi-fi, two rotating heads exclusively for hi-fi sound, in addition to the video heads, on the disc.
And, as shown in Fig. 2-2-7, audio FM signals are recorded to the deep part of the tape by the exclusive audio heads for hi-fi sound.
Then the video signals are recorded by the video heads. It is therefore necessary for the recording frequency (FM carrier) of the audio to reduce the luminance signal of the video portion to about 1/3 of the FM-modulated frequency and to widen the gap width.
If audio signals were to be recorded to the same depth (1.3 μm) as video signals, it would be necessary to record to a depth of 2.6 μm from the surface of the magnetic layer, and the gap width would be g = y/3.3, = 2.6/3.3 = 0.8μm. If the gap were made larger than this, recordings would be made to the deep part, but the playback output would decrease because the gap loss would increase during playback.
As a result, the optimum gap width of audio heads exclusively for hi-fi has been established to be 0.7 to 0.8 μm.
In order to eliminate crosstalk from the adjacent track and to eliminate the mix-in of video signals as a result of video signals being "piled" onto audio FM signals, the head gap azimuth angle (the inclination of the gap) is set to be ±30° for hi-fi audio heads and ±7°, the value prescribed for the Beta system.
This is because there is comparably less azimuth loss for low-frequency audio FM signals relative to the azimuth angle of ±7° of video signals in which the frequency bandwidth is video, and, moreover, because the manufacture of the head is difficult if the azimuth angle exceeds ±30°, and also because frequency changes occur under conditions of tracking deviation during playback or when playback output drops, and finally because DC drift (low-frequency distortion) is generated after demodulation.
For the above reasons, in addition to setting the azimuth angle of the hi-fi audio heads to be ±30°, installation is 45° from video head B, and the azimuth is set to be the opposite of that of video head B.
In addition, because there is no necessity for H arrangement correction of video signals by video head A, the hi-fi audio heads are installed 180° apart.

ARRANGEMENT OF AUDIO hi-fi HEADS

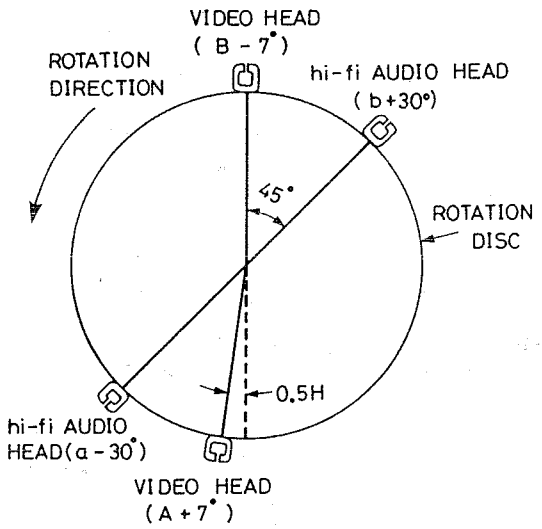


Fig. 2-2-6

CROSS-SECTIONAL VIEW OF TAPE IN WHICH VIDEO SIGNALS AND AUDIO FM SIGNALS FOR hi-fi VIDEO ARE RECORDED

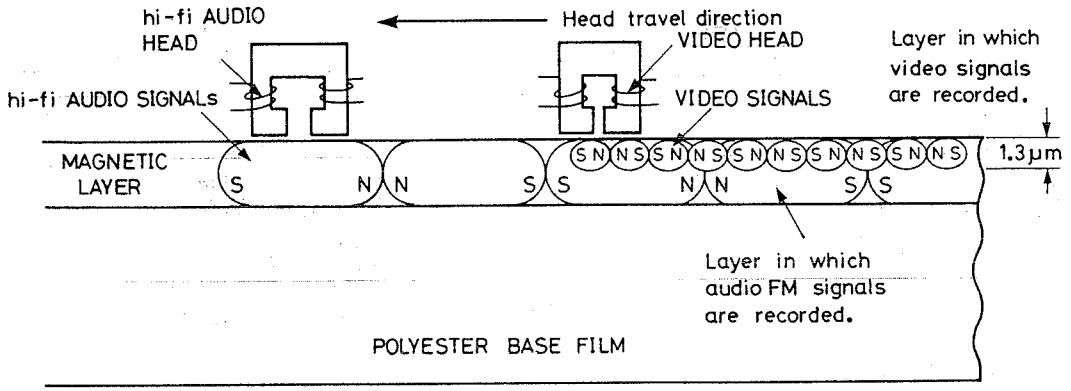


Fig. 2-2-7

3. PAL BETA hi-fi SYSTEM

(1) PAL (CCIR) SYSTEM AND NTSC SYSTEM

There is a great difference in the method used for signal processing for the recording and playback of audio FM signals for audio hi-fi of Beta type VCRs.

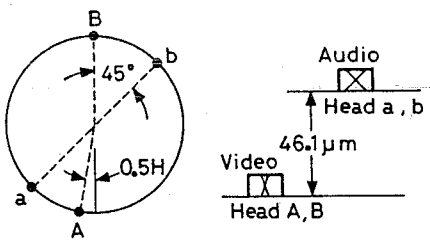
Audio signals are FM recorded by rotating heads for both systems, but there is a great difference in the multiplex recording method of the audio signals and the video signals onto the tape.

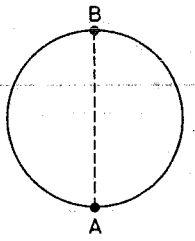
For the PAL system, as mentioned previously, two rotating audio heads exclusively for audio signals are used and these signals are recorded in the deep part of the tape, after which the video signals are recorded to the outer (shallow) layer of the tape by conventional video heads.

For the NTSC system on the other hand, the recording is made by inserting FM modulated audio signals into the spaces of the frequency spectrum of the FM modulated luminance signals and low-range conversion color signals. Thus, audio signals, after multiplexing to video signals, are supplied to conventional video heads and are recorded on the tape, with the result that there is no necessity for rotating heads exclusively for audio.

Consequently, the most obvious difference between the two systems is the construction of the heads: the PAL system has four heads whereas the NTSC system fundamentally had two heads.

Table 2-3-1 is a table of comparison of the PAL system and the NTSC system.

<table>
<tr><th></th><th>PAL (CCIR) audio FM deep recording</th><th>NTSC audio FM shallow recording</th></tr>
<tr><td>Head arrangement</td><td>4 Head
 

<table>
<tr><th></th><th colspan="2">Video Head</th><th colspan="2">A-FM Head</th></tr>
<tr><td></td><td>A</td><td>B</td><td>a</td><td>b</td></tr>
<tr><td>Azimuth</td><td>+7°</td><td>−7°</td><td>−30°</td><td>+30°</td></tr>
<tr><td>Gap</td><td colspan="2">0.4~0.5μm</td><td colspan="2">0.6~0.8μm</td></tr>
<tr><td>Head width</td><td colspan="2">43μm</td><td colspan="2">33μm</td></tr>
</table></td><td>2 Head
 

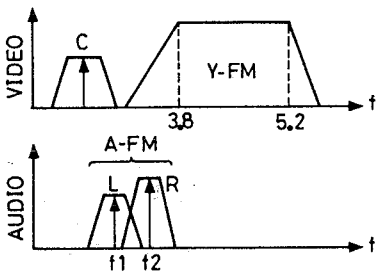
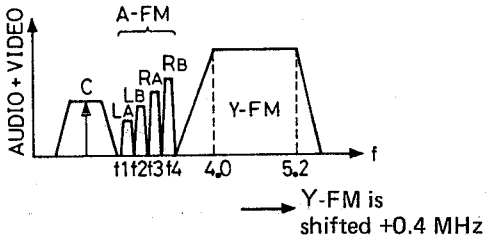
Video signals and audio FM signals are multiplex-recorded by video heads A and B.
<table>
<tr><th></th><th colspan="2">Video Head</th></tr>
<tr><td></td><td>A</td><td>B</td></tr>
<tr><td>Azimuth</td><td>+7°</td><td>−7°</td></tr>
<tr><td>Gap</td><td colspan="2">0.4~0.5μm</td></tr>
<tr><td>Head width</td><td colspan="2">33μm</td></tr>
</table></td></tr>
<tr><td>Recording method</td><td>Dual-write system</td><td>Frequency separation system</td></tr>
<tr><td>Frequency arrangement</td><td> 
</td><td> 
</td></tr>
<tr><td>Audio FM carrier frequency</td><td>f1 (L) = 1.44 MHz
f2 (R) = 2.1 MHz</td><td>f1 (LA) = 1.38 MHz
f2 (LB) = 1.53 MHz
f3 (RA) = 1.68 MHz
f4 (RB) = 1.83 MHz</td></tr>
<tr><td>Audio FM deviation</td><td>±200 kHz (Max)
±66.7 kHz (Standard level)</td><td>±75 kHz (Max)
±25 kHz (Standard level)</td></tr>
<tr><td>Recording level</td><td>Audio FM
f1 = 12 mA
f2 = 40 mA
Video
20 ~ 25 mAp-p</td><td>Luminance FM reference level ratio
Tolerance $^{+1}_{-5}$ dB
f1 = −26dB
f2 = −25dB
f3 = −24dB
f4 = −24dB</td></tr>
</table>

Table 2-3-1

(2) BASIC FORMAT AND PERFORMANCE OF PAL BETA hi-fi SYSTEM

A. Basic format of audio FM

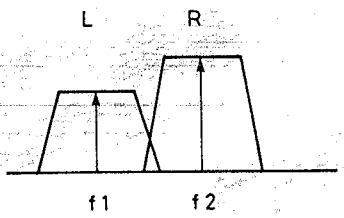
Item	Format
(a) Comparison of relationship of AF audio input/output Reference frequency level Line output level	 400Hz ± 66.7kHz dev. 33% modulation −10dBs
(b) FM carrier system	2-carrier system
(c) AF carrier frequency setting	f_1 (L) = 1.44MHz f_2 (R) = 2.10MHz
(d) AF carrier frequency arrangement	
(e) AF carrier frequency deviation volume	±200kHz (Center fix)
(f) AF carrier recording current characteristic	f_2 = Recording current (40 mA), maximum playback level of 2.1 MHz when video signals written on the another $f_1 = 0.3f_2$ (12mA)
(g) AF frequency precision	±20 kHz or less
(h) Audio FM modulation recording system	Positive modulation recording

Table 2-3-2

B. Audio FM basic performance

(1) Frequency response: 20 Hz – 20 kHz within $^{+1}_{-3}$ dB
(2) Distortion ratio: 0.5% or less (400 Hz, 33%)
(3) Dynamic range: 80 dB or more
(4) Crosstalk: 60 dB or more (400 Hz, 33%)
(5) Wow & flutter: 0.005% rms or less

(3) TAPE FORMAT

Figure 2-3-1 shows the tape format as a result of the use of audio hi-fi heads.

The most desireable arrangement is that in which there is the least possibility of the video tracks overlapping the three audio hi-fi tracks.

In order to satisfy this condition, the installation height difference of the audio hi-fi heads is established so that the center of the video heads in Fig. 2-3-1 will trace the center of the two audio FM tracks.

Figure 2-3-2 shows the relationship of the head installation height differences in order to satisfy the tape format conditions.

RELATIONSHIP OF HEIGHT DIFFERENCE OF AUDIO hi-fi HEADS AND VIDEO HEADS

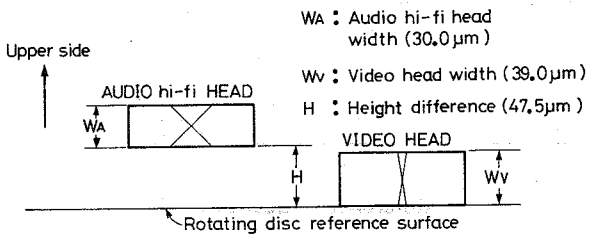


Fig. 2-3-2

TAPE FORMAT

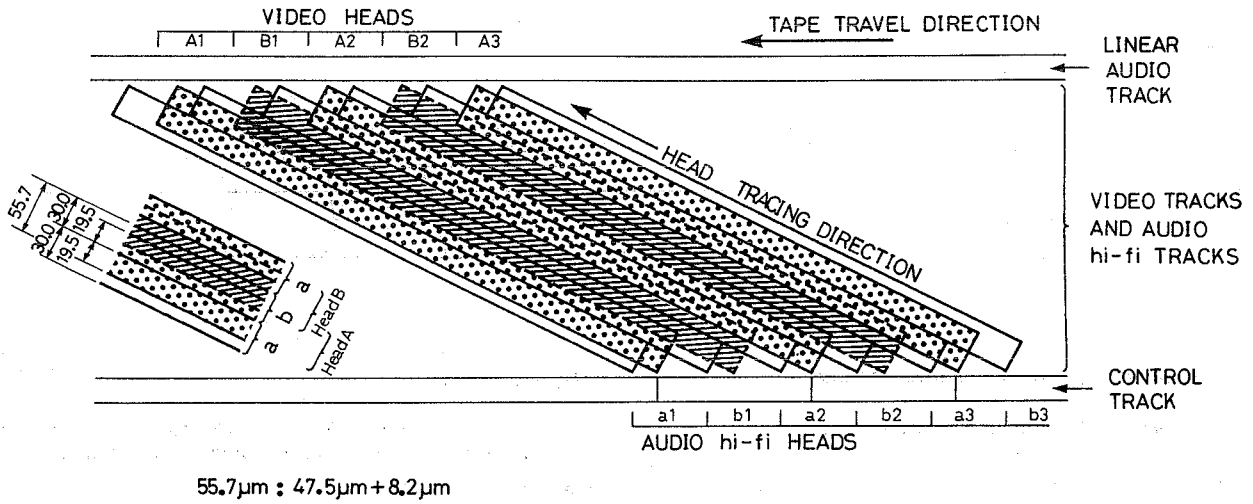


Fig. 2-3-1

(4) AUDIO FM CARRIER FREQUENCY

Figure 2-3-3 shows the frequency spectrum of audio signals and video signals.

The video signals are the same spectrum as ordinary Beta system VCRs, and to this are added the audio FM signals of the two channels for audio hi-fi use.

As a result, the luminance signals of the video signals are FM modulated, the white level being modulated to 5.2 MHz and the sync tip level (black level) being modulated to 3.8 MHz, and, as shown in the figure, the sideband waves of the luminance signal widen to the 1-MHz region. In addition, when chroma signals are recorded to the tape, they are converted to a low frequency in order to reduce the effects of "jitter" (caused by rotational irregularities of the video heads). For Beta system low-range conversion color signals, there is a color bus-carrier carrier frequency for low-range conversion to 688 kHz (44 times fH), and on that periphery there exists a frequency spectrum in the region of about ±400 kHz. For the audio FM signals, FM modulation is performed so that the center carrier for the left channel is 1.44 MHz and the center carrier for the right channel is 2.1 MHz. Because this modulation deviation has a wide band of ±200 kHz for each, there are the advantages of an excellent signal-to-noise ratio and minimized interference from video signals.

Next we will explain, referring to Fig. 2-3-4, the reason tha the audio FM signal's left channel (f_1 : 1.44 MHz) and right channel (f_2 :2.1 MHz) were established as the carrier frequencies.

1. The spurious component $(2f_1-f_2)$ should as far as possible not enter the audio FM band.
   $2(f_1 + \Delta f) - (f_2 - \Delta f) = 2f_1 - f_2 + 3\Delta f$
   This $2f_1 - f_2 + 3\Delta f$ does not enter $f_1 \pm \Delta f$ and $f_2 \pm \Delta f$, and in the same way
   $2(f_2 + \Delta f) - (f_1 - \Delta f) = 2f_2 - f_1 + 3\Delta f$
   This $2f_2 - f_1 + 3\Delta f$ does not enter $f_1 \pm \Delta f$ and $f_2 \pm \Delta f$.
2. This spurious level should be sufficiently low in relation to the level of the video signals (luminance and chroma signals).

3. The audio FM signal band should not enter the chroma signal band. This is because, for overwrite recording, "beat" interference is conspicuous relative to color on the TV screen during playback.
4. The setting of the bandpass filter for separation of $f_1 \pm \Delta f$ and $f_2 \pm \Delta f$ should be easy during playback. This is because separation is difficult if f_1 and f_2 are made to be close.
5. It should be possible to shift $3f_1 - 2f_2$ to outside the audible band.

FREQUENCY SPECTRUM RECORDED ON TAPE

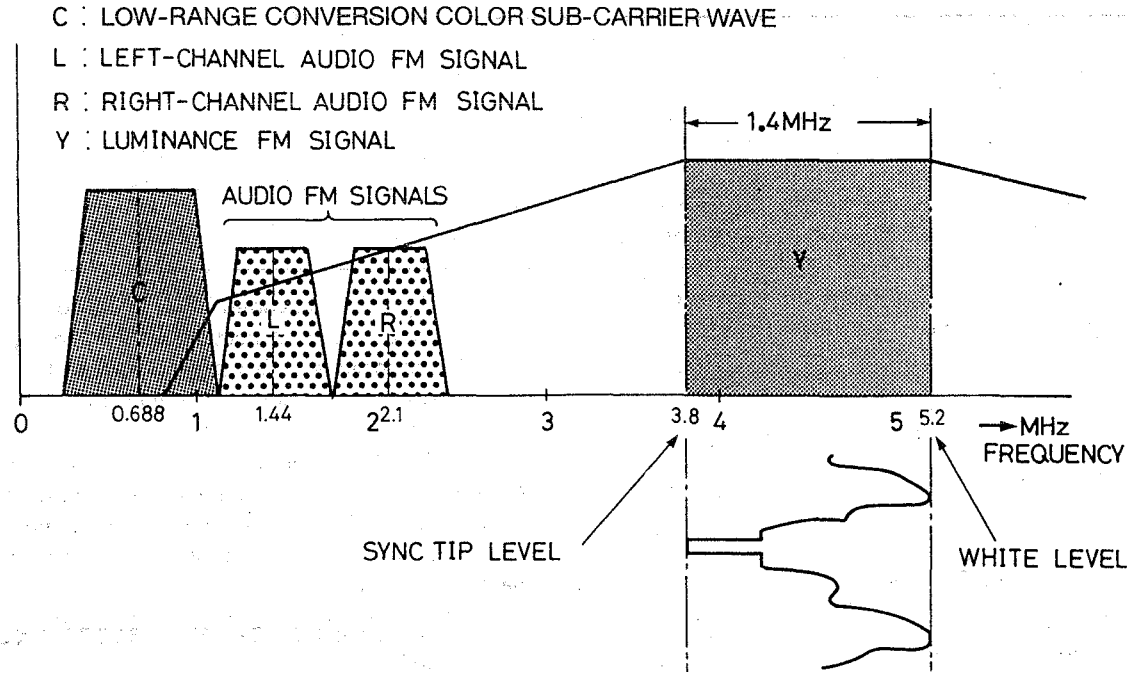


Fig. 2-3-3

AUDIO FM CARRIERS (f_1/f_2) AND SPURIOUS

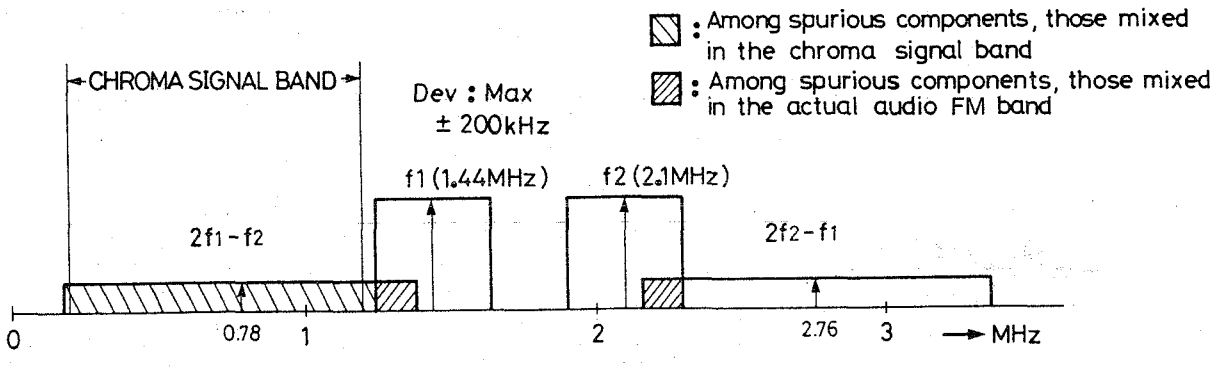


Fig. 2-3-4

(5) OPTIMUM RECORDING CURRENT

When determining the recording current of the audio hi-fi heads, the factors which must be considered are: a sufficient signal-to-noise ratio during playback, a large output level of f_1 and f_2, and that they are equivalent.

Consequently, the f_1 and f_2 recording currents are determined as described below, because audio FM signals are erased by the video recording current and the overall effects by frequency modulation.

The recording current of the audio FM signals (f_2) and the luminance (Y) signals is changed and the recording is made, and then the characteristics of the f_2 playback output level is as shown in Fig. 2-3-5.

In this figure, the ordinary PAL Beta system recording current of the luminance (Y) signal is 30 mA, but, because it is necessary to make a high setting of the f_2 playback output level, the recording current of the luminance (Y) signal is made small within the range in which the performance of the video signal can be maintained.

At this time, the recording current of the luminance (Y) signal is 22 mA, and the optimum recording current of f_2 is 40 mA.

In the same way, the relationship of the luminance (Y) signal and the audio FM signal (f_1) is shown in Fig. 2-3-6.

f2 RECORDING CURRENT/PLAYBACK OUTPUT CHARACTERISTICS

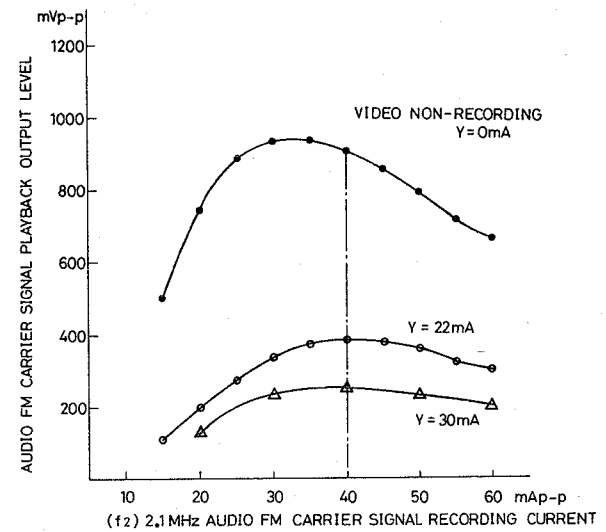


Fig. 2-3-5

f1 RECORDING CURRENT/PLAYBACK OUTPUT CHARACTERISTICS

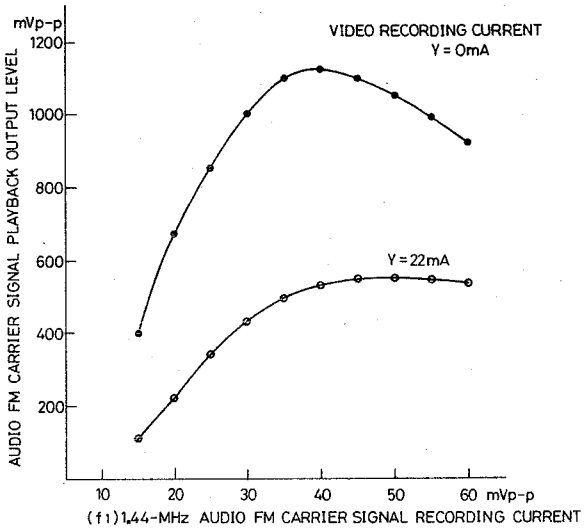


Fig. 2-3-6

Next the f_2 recording current is fixed at 40 mA, and each playback output level characteristic when the f_1 recording current is changed is shown in Fig. 2-3-7. In this figure, the point at which the f_1 and f_2 characteristics intersect is the optimum recording current of f_1.

In Fig. 2-3-7, the f_1 optimum recording current is 12 mA.

A described above then, the recording current of the video signals and the recording current of the audio FM signals are determined, and at the same time performance requirements are satisfied.

AUDIO FM CARRIER OPTIMUM RECORDING CURRENT

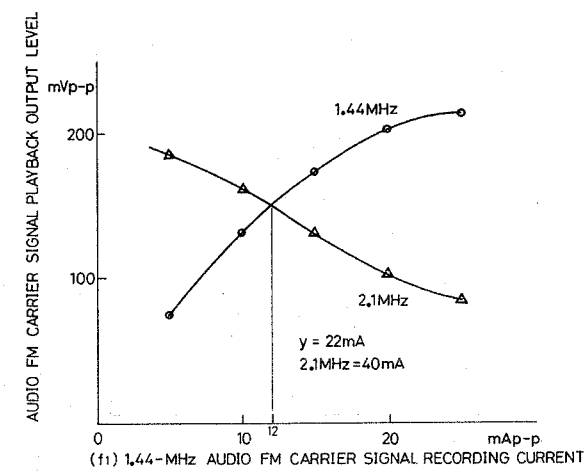


Fig. 2-3-7

(6) ROTARY TRANSFORMER

Transmission of signals from the video and audio circuits to the rotating heads is accomplished by a rotary transformer (rotor and stator) attached to the video head drum.

This method of signal transmission is the same as the method used previously. For Beta hi-fi however, "beat" interference is generated in the video signals when audio FM signals are mixed in the luminance (Y) signals. The detection limit of interference signals relative to luminance (Y) signals is approximately −45 dB. The relationship of the luminance (Y) signals and the f_1 and f_2 recording current levels is shown in Fig. 2-3-8. Because, in Fig. 2-3-8, the f_2 level is high and moreover the carrier frequency is also near the luminance (Y) signal, interference by f_2 becomes a problem. This interference is mainly interference caused by crosstalk of the rotary transformer. In addition, crosstalk of f_2 and of the luminance (Y) signal reciprocally become bias recording. For this reason, f_2 is boosted further, and, for the allowable value of video signal and audio FM signal crosstalk, as shown in Fig. 2-3-9, a level of 61 dB or higher is necessary.

f_1, f_2 AND LUMINANCE (Y) SIGNAL RECORDING CURRENTS

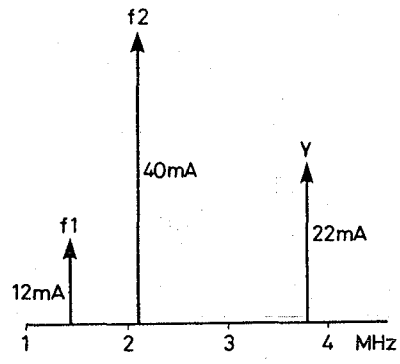


Fig. 2-3-8

ALLOWABLE CROSSTALK VALUES OF VIDEO SIGNALS AND AUDIO FM SIGNALS DURING RECORDING

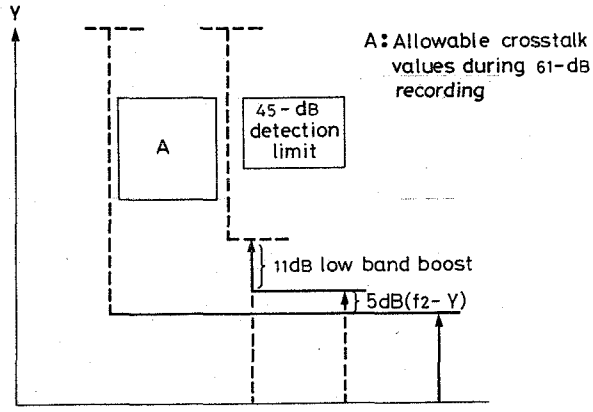


Fig. 2-3-9

In order then to satisfy the conditions described above, it would be advantageous to have a rotary transformer exclusively for audio hi-fi signals, installed at a different location than that for video signals, but this would necessitate extensive changes in the construction of the video head drum, thus making it difficult to realize in actual practice.

Thus, by using the short ring shown in Fig. 2-3-10, a rotary transformer with a unified construction for video signals and for audio FM signals is adopted.

By following this method, the crosstalk level between video signals and audio signals can be suppressed to 70 dB.

ROTARY TRANSFORMER FOR PAL BETA hi-fi SYSTEM

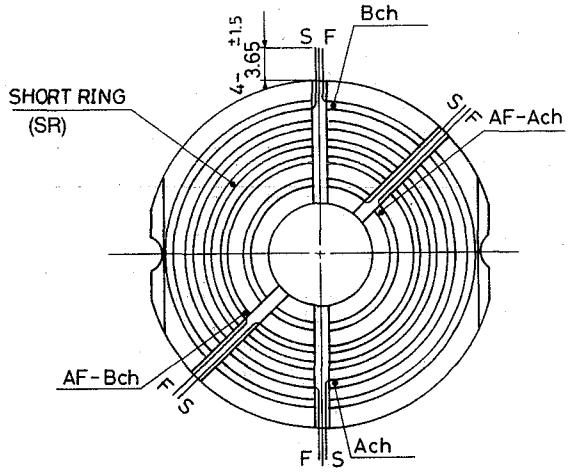


Fig. 2-3-10

4. AUDIO FM CIRCUITRY

(1) OUTLINE AND FEATURES OF AUDIO FM CIRCUITRY

The features of the audio FM circuitry are mainly in the playback system, and will be explained based upon the block diagram in Fig. 2-4-1.

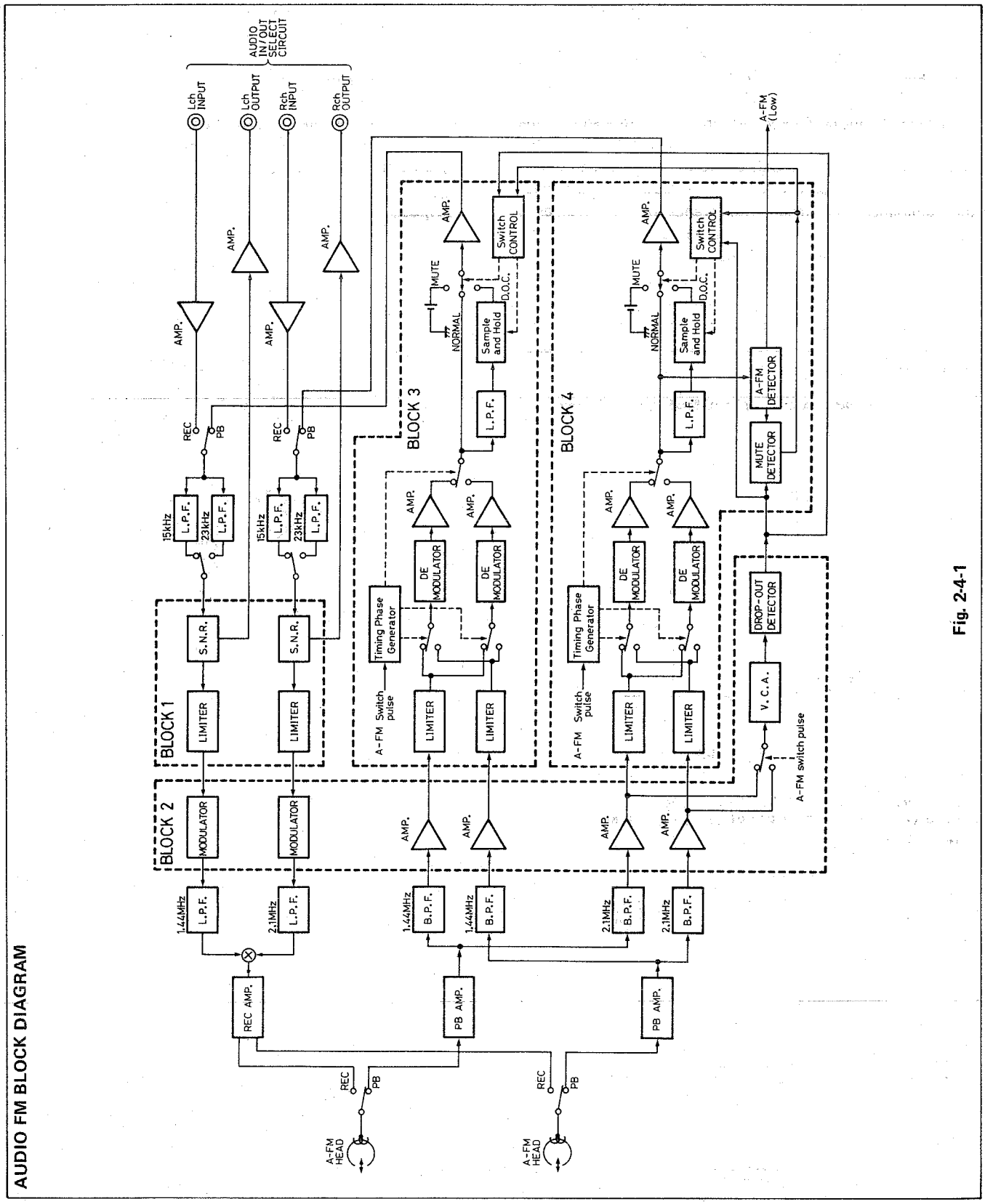


Fig. 2-4-1

The features of the audio FM circuitry are in four of the blocks in Fig. 2-4-1, and these blocks can be classified as follows.

1. Block 1
   SNR (Silent Noise Reduction)
2. Block 2
   FM modulator, playback RF amplifier, and drop-out detector
3. Block 3
   Left channel 2-circuit demodulator
4. Block 4
   Right channel 2-circuit demodulator, drop-out compensation and muting circuit, and A-FM discrimination circuit

(2) BLOCK 1

Block 1 is a logarithmic compression-and-expansion type of noise-reduction system; this SNR (Silent Noise Reduction) system consists of the V.C.A. (voltage control amplifier) and the M.O.A. (main operational amplifier), as well as the detector, over-modulation limitter, overshoot limitter, etc.

The following is an explanation of the noise-reduction system.

Because an FM system is used, in the same way as for luminance signals, in which the audio signal modulation method is used for audio hi-fi, delta noise is generated during demodulation.

In order to reduce this delta noise, usually an emphasis circuit is used, but, because the noise reduction effect is not sufficient by using it only, a wide dynamic range cannot be obtained.

Because, for frequency modulation, there is almost no level fluctuation of the audio signal output even though the level of the playback carrier wave is varied, there is no problem even if there is compression and expansion of great magnitude.

It is for this reason that noise-reduction circuitry in which emphasis (dynamic emphasis) which functions from low frequencies to high frequencies and circuitry in which signal compression and expansion are combined is used.

In this way, not only can delta noise be reduced, but, at the same time, a drop in linearity at high frequency is also prevented, and, moreover, noise at low frequency is also at the same time reduced.

A. Frequency modulation system delta noise

The FM system is a modulation system which causes frequency deviation is proportion to input signals; the relationship is frequency deviation = modulation index x modulation frequency; the modulation index becomes proportional to the magnitude of the noise component and the carrier wave, and, because input noise has a flat frequency spectrum, the result is uniform, as in (a) of Fig. 2-4-2.

In addition, because the modulation index can be considered to be the frequency of the noise after demodulation, the frequency deviation is changed in proportion to that.

As a result, the output noise of the FM demodulator is made proportional to the frequency, so that the frequency response increases in size, becoming a triangle-shaped noise characteristic, as shown in (b) of Fig. 2-4-2.

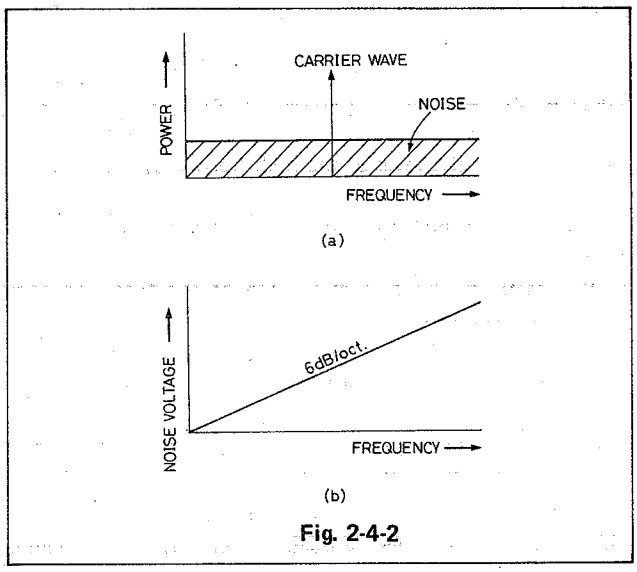


Fig. 2-4-2

B. Reduction of delta noise

In general, the frequency spectrum of music has a tendency to decrease (the high-range component to become less) as the frequency becomes higher.

Thus, taking advantage of this characteristic, the fixed emphasis of 50 to 75 μsec. shown in Fig. 2-4-3 is usually employed.

However, with today's wide diversification of recording media, synthesizer music, etc., and the great range of high-range signals, high-range linearity decreases at fixed emphasis.

For that reason then, and in order to improve linearity and reduce delta noise, a variable emphasis for changing the amount of emphasis in accordance with input level and frequency was newly developed, thus improving the dynamic range.

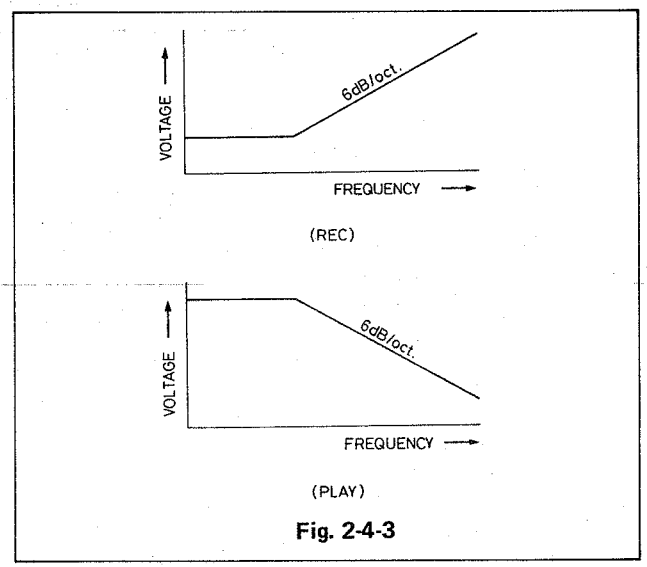


Fig. 2-4-3

C Noise-reduction system

The dynamic range, including FM modulation and demodulation and the tape system, and considering the frequency response to be flat, is about 50 dB.

This is because delta noise, a particular characteristic of frequency modulation, and noise from the tape system are included.

The method generally employed to improve the dynamic range of frequency modulation is by using emphasis circuitry to increase the level of the high-frequency region of the frequency response during recording, and then, during playback, by using de-emphasis circuitry to attenuate the high-frequency area.

In this method, the dynamic range of the high-frequency area is not expanded.

It was for this reason, therefore, that a noise-reduction system for audio hi-fi use was developed, in order to greatly expand the dynamic range.

The noise-reduction system employed in the PAL Beta system is called the SNR system.

This SNR system functions to apply emphasis-1, 2 and 3 in three steps to the high-frequency area and low-frequency area centered upon 400 Hz, the audio frequency band. Then a voltage-control amplifier is used for compression-expansion of the signal.

There are, in addition to the SNR system, the following noise reduction systems:

- Dolby B system

By applying one-step emphasis to the audio signal frequency band, the major portion of the high-frequency area is compressed and expanded.

- Dolby C system

By applying two-step the high-frequency area is compressed and expanded.

- dbx system

Linear compression-expansion at 1/2 compression ratio (logarithmic) to entire audio signal frequency band.

- Beta noise-reduction system

Compression-expansion of most ranges by one-step emphasis.

D. SNR system

The SNR system is composed of waiting circuitry, detector circuitry, V.C.A. circuitry, etc. which function to compress and expand, in accordance with the frequency and sound level, the sound signal and the emphasis for reduction of delta noise (which spreads to the low-frequency and high-frequency areas), centered upon 400 Hz, which is generated during FM conversion.

This SNR system can be thought of, moreover, as being composed of two parts for purposes of its functions: the main op. amplifier and the two expanders, which are composed of the other blocks.

In Fig. 2-4-4, the OPEN/CLOSE switch terminal is shown in the OPEN condition. When this OPEN/CLOSE switch is in the OPEN condition, the main op. amplifier functions as a voltage follower (gain = buffer amplifier 1), and obtains the decoding characteristic from EXPANDER OUT.

As a result therefore, only the EXPANDER portion of the SNR system functions during tape playback.

During recording, the OPEN/CLOSE switch changes to the CLOSED condition, and, by inserting the decoder circuitry consisting of the EXPANDER into the negative-feedback loop of the main op. amplifier, the encode characteristic is obtained from COMPRESSOR OUT of the SNR system.

SNR SYSTEM BLOCK DIAGRAM

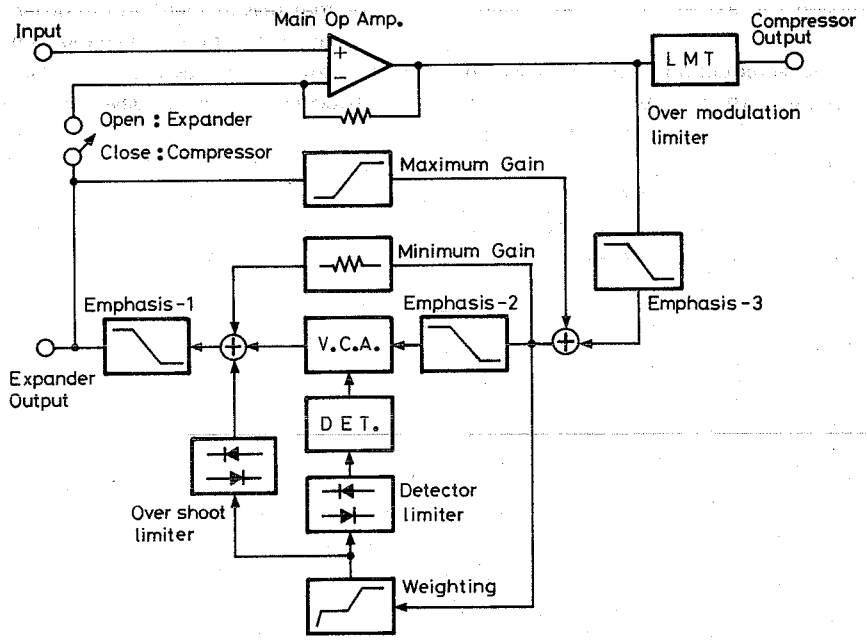


Fig. 2-4-4

The function of each block in Fig. 2-4-4 is as described blow.

- V.C.A.

This is the voltage control amplifier. Gain becomes large when the input signal level is high, and becomes small when the input signal level is low.

- EMPHASIS

Emphasizes a particular frequency according to the CR time constant.

- WEIGHTING

By attenuating the low-range frequency by the CR time constant, the result is a frequency response with a raised high area.

- MAXIMUM GAIN

By determining the maximum gain of the audio signals when there is compression (during recording) and expansion (during playback), excessive V.C.A. level is regulated.

- MINIMUM GAIN

By determining the low audio signal level which cannot be controlled by the V.C.A., V.C.A. negative gain is prevented.

- LIMITER

Lowers the encode amount during high level.

The following are the CR time-constant reference values of the SNR system.

• EMPHASIS-1	77.4μsec (2.06kHz),	18.0μsec (8.82kHz)
• EMPHASIS-2	158μsec (1.01kHz),	35.3μsec (4.51kHz)
• EMPHASIS-3	186μsec (0.856kHz),	150μsec (1.06kHz)
	187μsec (0.850kHz),	150μsec (1.06kHz)
• MAX. GAIN PASS	300μsec (0.531kHz),	150μsec (1.06kHz)
• WEIGHTING	1.34msec (119Hz),	158μsec (1.00kHz)
	12.2μsec (13.0kHz),	39.0μsec (4.08kHz)

The theoretical values of noise attenuation by the SNR system are as follows.

100Hz	1kHz	10kHz
30.0dB	30.5dB	41.5dB

Figure 2-4-5 shows the extent of the expansion of the dynamic range when signals are compressed during recording and then expanded during playback.

The dynamic range for FM recording and playback is approximately 50 dBs, which is the range between the maximum value (−10 dBs) of the audio signal, as determined by the maximum frequency deviation (±200 kHz) of the frequency modulation, and the noise level (−60 dBs).

In order to expand this dynamic range to about 80 dB, the procedure described below is followed.

In the FM recording/playback system, low level signals (A) buried in noise are amplified to (B) level during recording and recorded at that level.

During playback, B-level signals are played back as (B'), and these (B') signals are attenuated to the original (C) level (A = C).

Signals of a high signal level (D) are, however, amplified almost not at all during recording, and are recorded at (E) level. Then, during playback, these E-level signals are played back as (E'), but these (E') signals are attenuated almost not at all, and return to the original (F) signal level (D = F).

In this way, then, the dynamic range is expanded.

DYNAMIC RANGE OF FM RECORDING/PLAYBACK

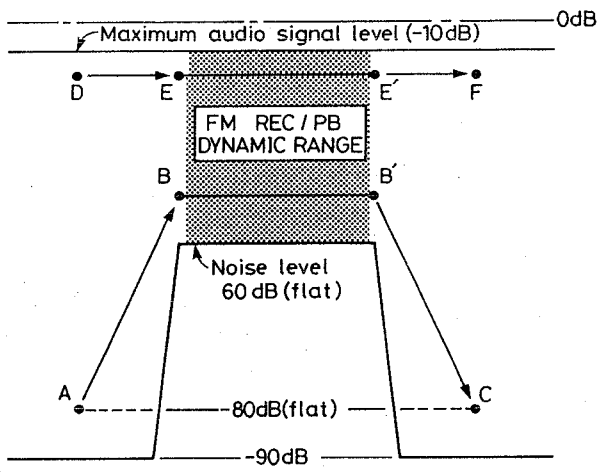


Fig. 2-4-5

In Fig. 2-4-6, the most important parts of the EXPANDER have been taken out. (In this figure therefore, LMT, EMPHASIS, LIMITER, MINIMUM GAIN and MAXIMUM GAIN have been omitted.)
Because the OPEN/CLOSE switch shown in Fig. 2-4-4 is in the OPEN condition during playback, input signals are, as shown in Fig. 2-4-7, input to the EXPANDER.
Signals input to the EXPANDER are input to WEIGHTING as well as to the V.C.A. After "breathing" (a particular phenomenon which is characteristic of noise reduction by compression and expansion) is reduced at WEIGHTING, the signals pass through the DETECTOR and are input to the V.C.A., but signals produced by the loop composed of EXPANDER input to weighting → DETECTOR → V.C.A. become the CONTROL IN characteristic shown in Fig. 2-4-8. When the CONTROL IN is large, the V.C.A. functions to increase gain, and, when CONTROL IN is small, functions to decrease gain.
As a result of this function of the V.C.A., low-level signals, as shown by (B') in Fig. 2-4-5, become signals of even smaller (C) level, and the high-level signals (E') become signals of an almost unchanged level (F).

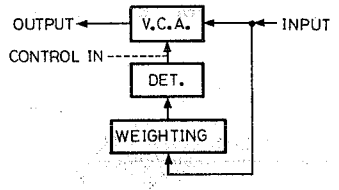


Fig. 2-4-6

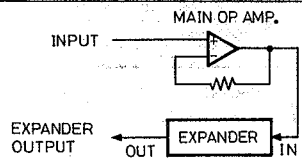


Fig. 2-4-7

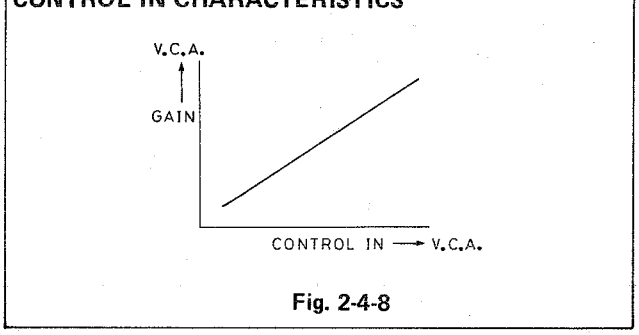


Fig. 2-4-8

The gain of the V.C.A. changes according to the CONTROL IN signal level, and the audio signal attenuation relationship is as follows.

- When the signal level of the audio playback input is great, V.C.A. gain becomes large, because the control signal level is high, and the amount of attenuation of the audio signal becomes small.
- When the signal level of the audio playback input is small, V.C.A. gain becomes small, because the control signal level is low, and the amount of attenuation of the audio signal becomes large.

Consequently, because there is almost no attenuation when the signal level of audio playback input is great, and there is a great amount of attenuation when the signal level of audio playback input is small, the audio signal is therefore expanded during playback.

Because the OPEN/CLOSE switch shown in Fig. 2-4-4 is in the CLOSED condition during recording, input signals enter the EXPANDER feedback circuit of the MAIN OP. AMPLIFIER, as shown in Fig. 2-4-9. Because it in this way becomes a feedback loop to the negative input of the MAIN OP, AMPLIFIER, the audio input signals are compressed, and the compressor output characteristic shown in Fig. 2-4-10 becomes the reverse of the CONTROL IN characteristic shown in Fig. 2-4-8.
In addition, the expander output shown in Fig. 2-4-9 becomes the EE monitor during recording. The relationship of audio signal levels during recording is as described below.

- When the audio signal level is large input, the amount of nagative feedback to the MAIN OP. AMPLIFIER becomes large, and, because the gain of the MAIN OP. AMPLIFIER becomes small, the amount of attenuation of audio signals becomes great.
- When the audio signal level is small input, the amount of negative feedback to the MAIN OP. AMPLIFIER becomes small, and because the gain of the MAIN OP. AMPLIFIER becomes large, the amount of attenuation of audio signals becomes small.

As a result, attenuation is great when, during recording, the audio signal level is great, and, because there is almost no attenuation when the audio signal level is small, the audio signals are compressed during recording.

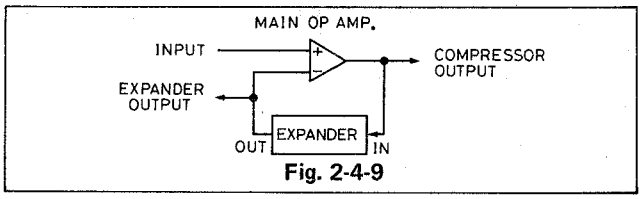


Fig. 2-4-9

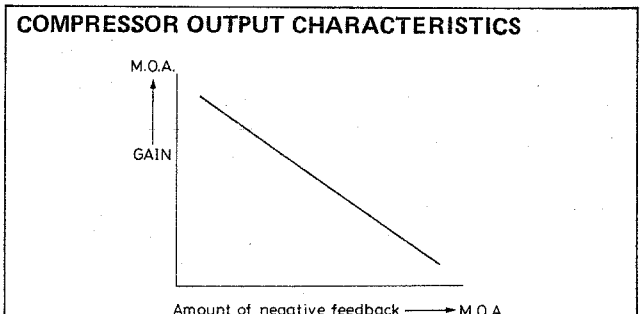


Fig. 2-4-10

(3) BLOCK 2

Block 2 is composed of the FM modulators, playback RF amplifiers, and drop-out detectors.

Figure 2-4-11 shows the block diagram of the modulator system. In this system, the FM modulators function to modulate the frequency deviation ±200 kHz linearly and to improve the stability of the oscillation level and oscillation frequency.

Figure 2-4-12 shows the block diagram of the playback RF amplifiers and the drop-out detectors.

The RF amplifiers are the pre-amplifiers for the demodulators for signal processing (blocks 3 and 4). The drop-out detectors, by envelope detection of the absolute demodulation limit value, hold the V.C.A. amplifier playback level to the prescribed level in order to absorb any variations of the pre-amplifier or the head assembly especially for tape and audio hi-fi use.

MODULATOR SYSTEM BLOCK DIAGRAM

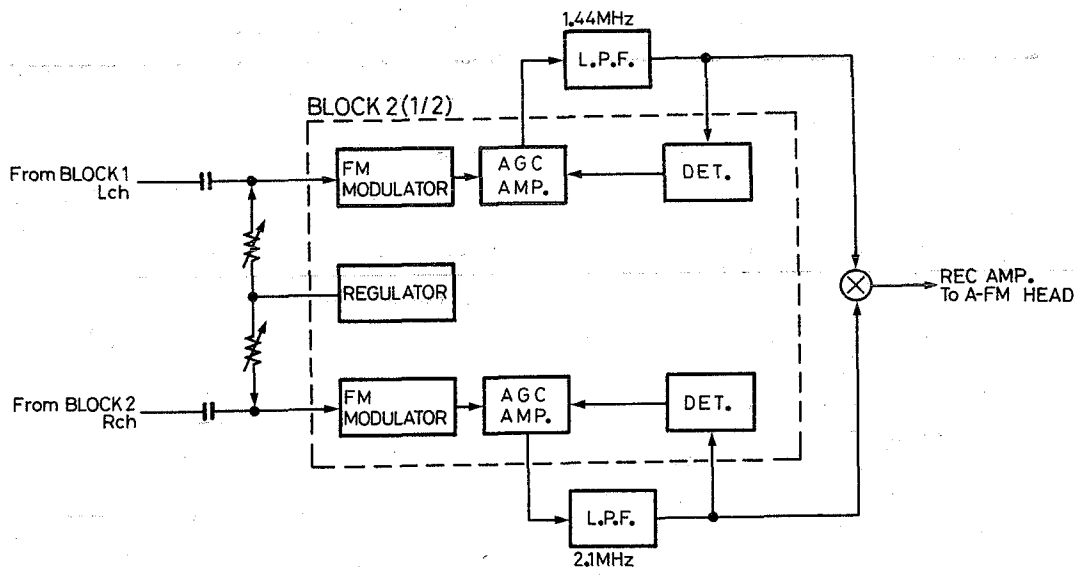


Fig. 2-4-11

RF AMPLIFIER AND D.O.C. DETECTOR BLOCK DIAGRAM

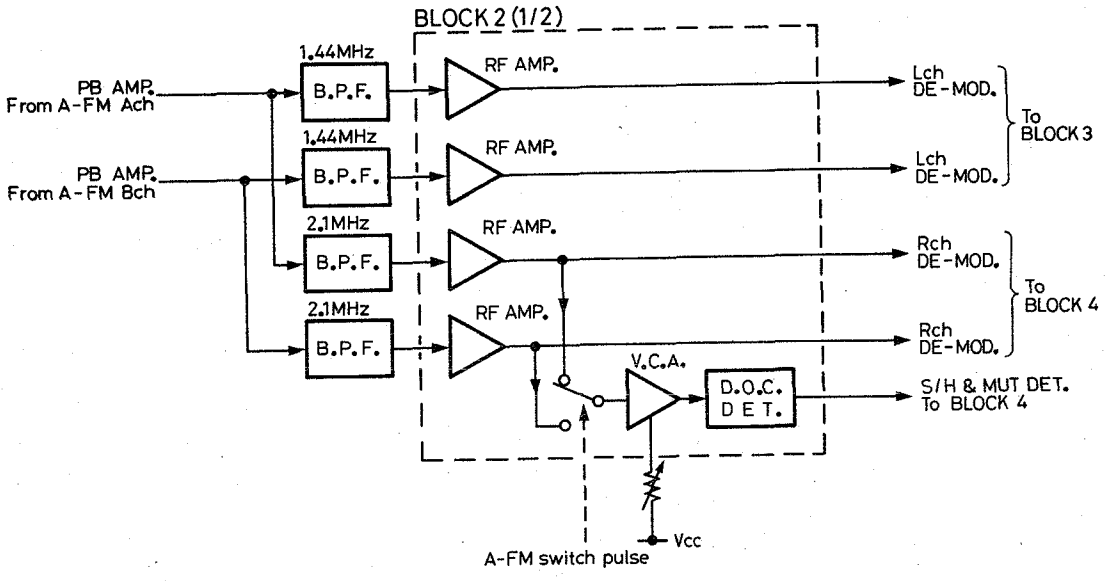


Fig. 2-4-12

(4) BLOCK 3

Block 3 is composed of the left channel 2-circuit demodulator, and this circuit is also used in block 4 for the right channel 2-circuit demodulator.

The following is an explanation of switching noise, which is the biggest problem resulting from the adoption of audio hi-fi by the rotary-head system from the conventional laterally positioned audio head.

A. hi-fi video switching noise

For video recording, the video signal for one field is recorded diagonally onto the tape by the rotating head, or, to express it in another way, the signal to be recorded is "cut" every 1/25 of a second and is recorded on the tape.

There are, in these video signals, a vertical blanking period for each field. Because there is a time (the retrace line erase period) in which the screen is momentarily blank, the switching of these signals is virtually no problem.

For audio hi-fi VCR however, these audio signals, which were originally supposed to be continuous, are cut each 1/25 of a second and recorded on the tape, and, when the tape is played back, they are converted back to continuous signals.

For video signal recording and playback, one of the two heads contacts the tape first, passes along the tape, and then, immediately prior to its breaking contact with the tape, the second head makes contact.

A certain impact "shock" is caused at the moment this second head contacts the tape, and the video bends at the top part of the screen, causing the skew distortion phenomenon.

Actually, this skew distortion is concealed within the mask, and so is not noticed.

Because the audio signals of hi-fi video are recorded and played back by rotary heads, noise does occur as a result of the "connection joints", and this is called switching noise.

As can be seen in Fig. 2-4-13, there is multiplication of noise in the 1-kHz region of the frequency spectrum.

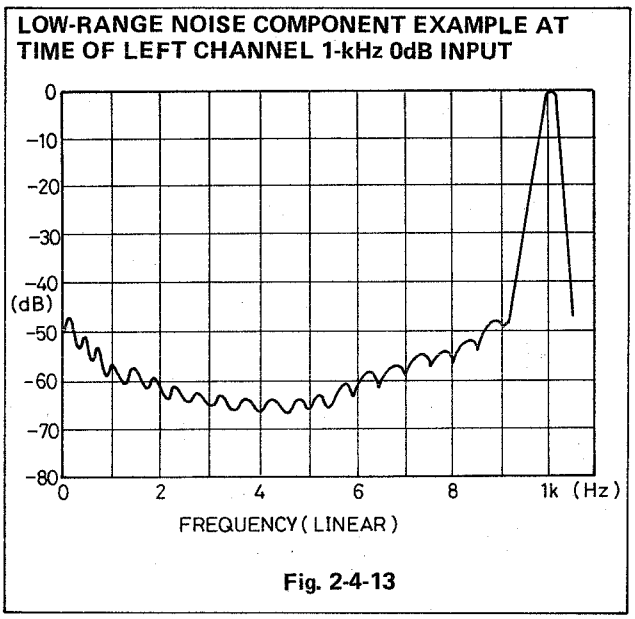


Fig. 2-4-13

B. Principle of processing switching noise caused by helical scanning "connection joints'

For audio hi-fi VCR, because audio signals are FM modulated and are switched to multiple (A channel and B channel) heads in order, thus overlapping on the tape for recording and playback, the signals during playback become the intermittent signals shown in Fig. 2-4-14, and transient noise is generated in the audio signals in modulation output.

The fundamental principle for elimination of this noise which occurs during head switching is the use of the overlap recording part shown in Fig. 2-4-15.

In the matrix circuit composed of S1, S2 and S3, the noise where the signal is lacking at S1 is compensated for by the other (S2), and vice-versa.

Then, the signals which have passed through S1 and S2 are each detected, respectively, by the two demodulators, and, by switching at S3 at the part where there is no signal lack, it is possible to take out signals which are continuous and without "connection joints" and noise.

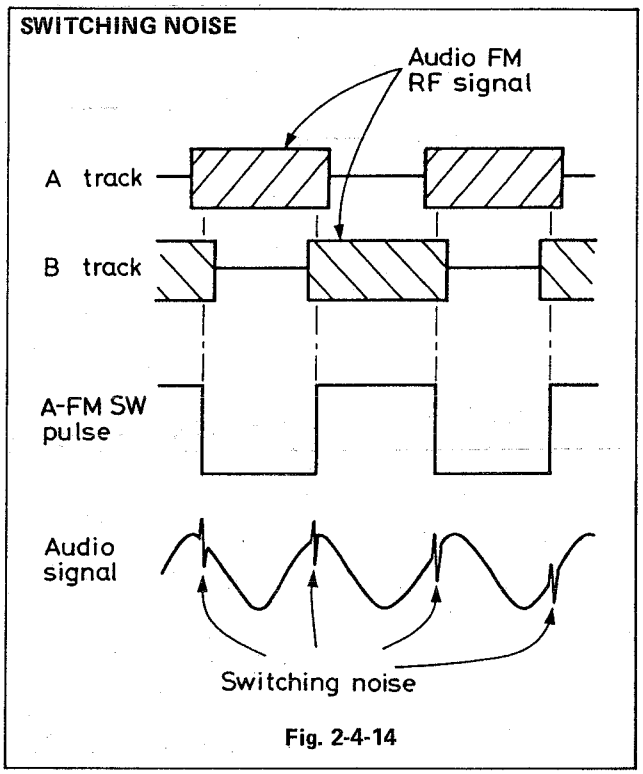


Fig. 2-4-14

PRINCIPLE OF AUDIO FM SYSTEM SWITCHING NOISE PROCESSING

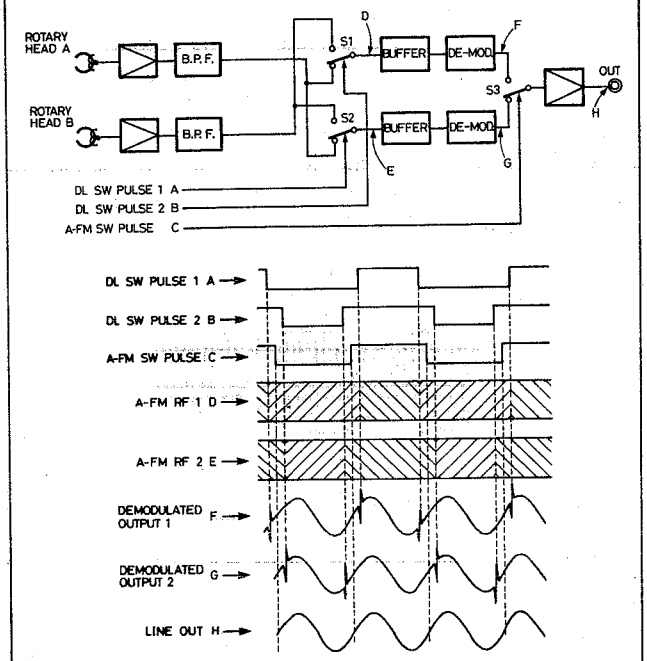


Fig. 2-4-15

In addition, the phase of playback signals is not perfectly uniform at the "connection joints" of the output of the two audio heads, because of differences in tape tension, etc. For that reason, as shown in Fig. 2-4-16, the phase of the signal of the switched portion is changed, causing the generation of a pulse-like noise which corresponds to the changed phase in the demodulation output.

Because there is, for video signals, a period of vertical blanking for head switching, it is not visibly noticeable; for audio signals however, this would, if not corrected, become significant interference. The fundamental circuitry used for reduction of this switching noise is the circuit called the "sampling-hold" circuit shown in Fig. 2-4-16.

In this sampling-hold method, there is a sampling and hold (of the value of the demodulation signal, including pulse noise, immediately prior to the noise) by the hold pulse produced from the head-switching pulse, with the result that there is no noise in the output.

There are various conceivable methods of using this fundamental principle for the circuit for reduction of switching noise, and its use reduces switching noise to a great degree.

SAMPLING-HOLD METHOD BY HEAD SWITCHING

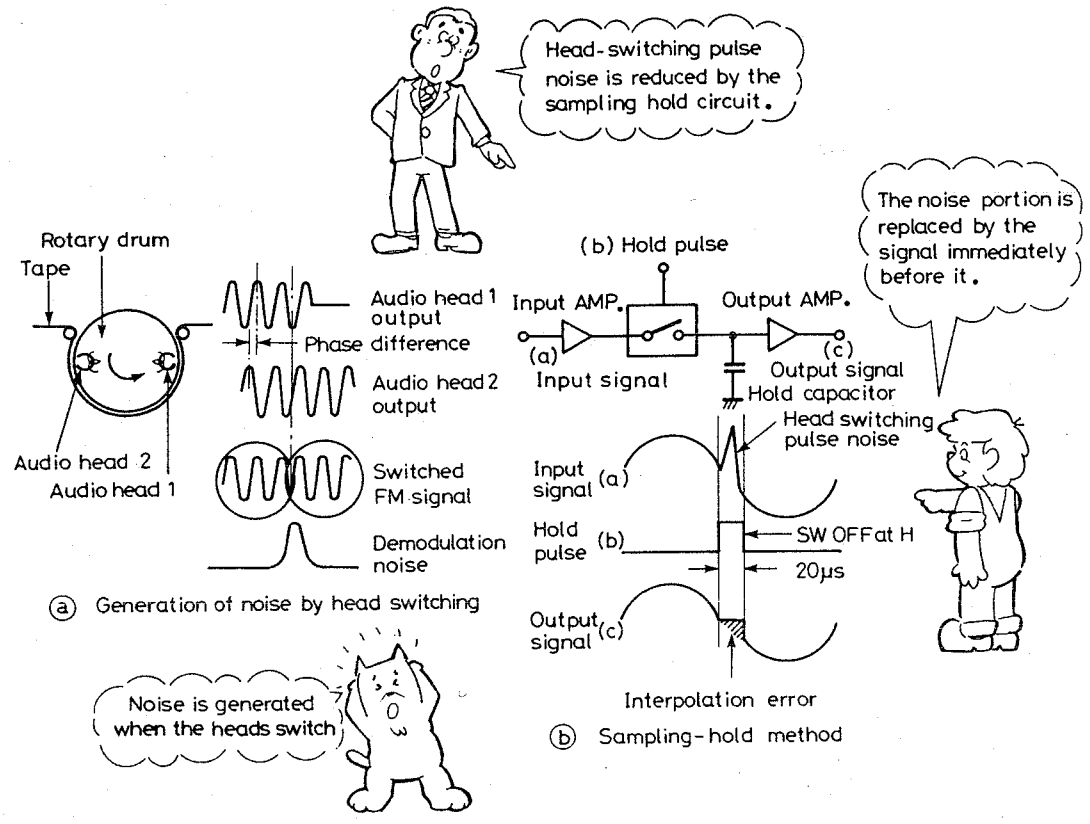


Fig. 2-4-16

C. Disadvantages of switching noise processing by preliminary hold.

The following is an explanation of the principles of demodulation for the PAL Beta system.

When, in Fig. 2-4-17, the RF playback output of the audio FM signal is switched (channel A and channel B) and demodulated, transient noise (Fig. a) occurs immediately after the switching.

In order therefore to suppress this noise, preliminary hold for t time is used, and it becomes as shown in Fig. b.

SWITCHING NOISE BY PRELIMINARY HOLD

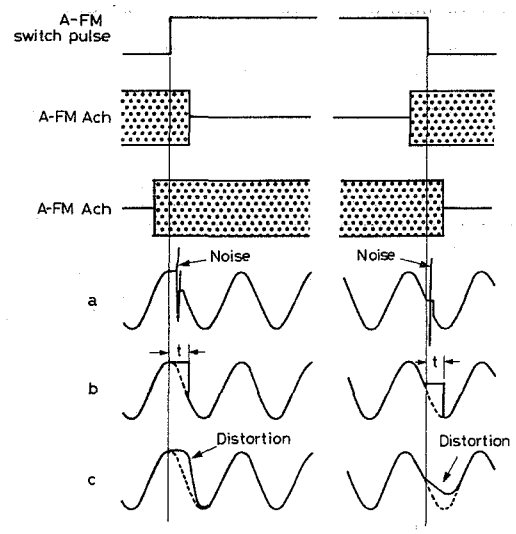


Fig. 2-4-17

And, after the signal passes through the low-pass filter, the waveform shows the distortion shown in Fig. c. When this is further expanded by noise reduction, switching noise such as shown in Fig. 2-4-18 is generated in the output.

For playback of an ordinary music source, there is no problem, from the audible standpoint, regarding noise by preliminary hold up to about 200 μsec. This preliminary hold time is well below the limit of detection, because the time becomes about 7 to 10 μsec. thanks to improvements of the low-pass filter, bandpass filter, etc.

SWITCHING NOISE BY PRELIMINARY HOLD

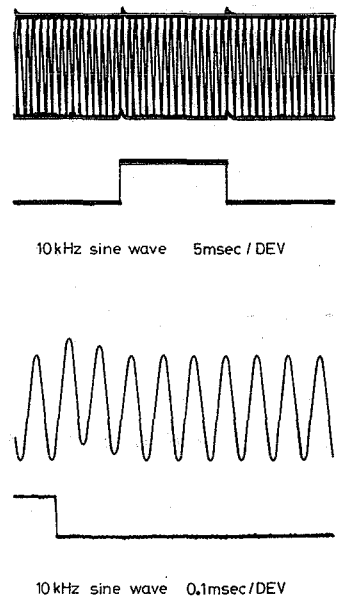


Fig. 2-4-18

D. Principles of the 1-channel, 2-circuit demodulator

Figure 2-4-19 shows the block diagram of the 1-channel, 2-circuit demodulator.

This demodulator is designed to place importance upon the audio FM signal's linearity and signal-to-noise ratio, and, because the carrier frequencies are comparatively low (1.44 MHz and 2.1 MHz), the pulse-count type of FM demodulator is used.

TWO-DEMODULATOR/1-CHANNEL PRINCIPLE

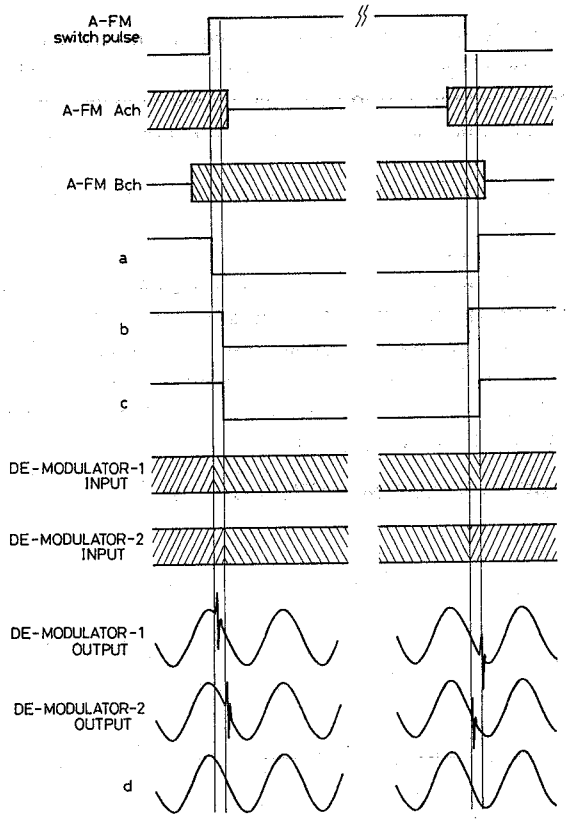


Fig. 2-4-20

As can be seen in Fig. 2-4-20, the transient noise which is generated at the time of A-FM head switching is generated immediately after the switchover, and the RF channel A and RF channel B signals are recorded in an overlapped condition.

By taking advantage of this, delay pulses in (a), (b) and (c) are produced from the A-FM switching pulse by the T.P.G. (Time Phase Generator).

Then, by switching the input to the demodulator by this delay pulse, and by further switching the portion of no noise by the playback output from the demodulator, it is possible to obtain playback output, as shown in (d), with absolutely no switching noise.

Figure 2-4-21 shows 10-kHz playback output which used the 1-channel, 2-circuit FM demodulator, and, as can be seen, switching noise is almost entirely not visible.

TWO-DEMODULATOR/1-CHANNEL PLAYBACK WAVEFORM

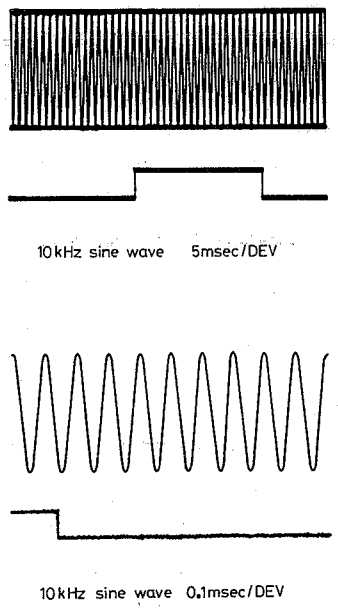


Fig. 2-4-21

TWO-DEMODULATOR/1-CHANNEL BLOCK DIAGRAM

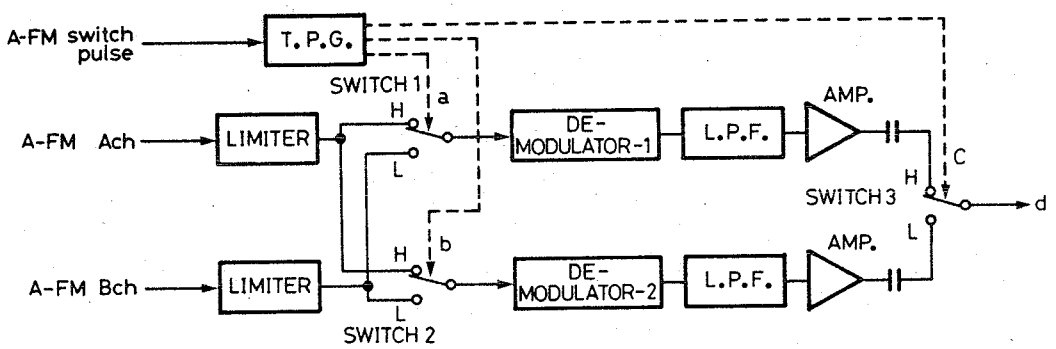


Fig. 2-4-19

(5) BLOCK 4

Block 4 is composed of the right channel 2-circuit demodulator, drop-out compensation and muting circuitry, and audio FM discrimination circuitry. Of these, the explanation of the right channel 2-circuit demodulator will be omitted, because it is the same as for the left channel 2-circuit demodulator explained above.

A. Drop-out compensation and muting circuitry

Usually, when drop-out occurs RF envelope output completely disappears. For the deep recording method however, when drop-out occurs on the surface layer (on which video signals are recorded), the RF envelope of the audio FM signals increases.

In this way, then, for shallow drop-out the RF envelope level is in the direction of increase, so that there is therefore no necessity for compensation. As a result, drop-out compensation is made to reach the deep layer of the tape where audio FM signals are recorded.

As for the method of compensation, for the 200 to 300-μsec. drop-out which does not become audible noise, a preliminary hold method (which is different from the preliminary hold noise-processing method used up to 200 μsec.) is used.

This preliminary hold is accomplished by the S/H (sample and hold) circuitry after signals have passed through the low-pass filter, so that audio FM signals will not be applied to the noise component.

For long drop-out which exceeds 300 μsec., processing is in two steps by muting the audio FM signals.

The muting process is set, moreover, to a fixed reset time (approx. 100 msec.) so that there will not be repeated ON/OFF operation after drop-out has finished. Figure 2-4-22 shows the drop-out compensation and muting circuitry block diagram.

B. Audio FM discrimination circuitry

The audio FM discrimination circuitry is signal-production circuitry which functions to switch to the normal audio circuitry if playback of audio FM signals is impossible because there is a serious scratch on the tape or if there are no audio FM signals recorded on the portion of the tape being played back.

For the usual audio hi-fi system, this circuitry discriminates the existence or not of the audio FM signal carrier.

For the audio hi-fi of the PAL system, the demodulation output of the audio FM signal is compared to the normal audio signal, and this discrimination is made according to the level of the high-range noise, by the increase of approximately 20dB of this high-range noise level.

In this way it is possible to make a more appropriate determination relative to demodulation output.

DROP-OUT COMPENSATION AND MUTING CIRCUITRY AND AUDIO FM DISCRIMINATION CIRCUITRY

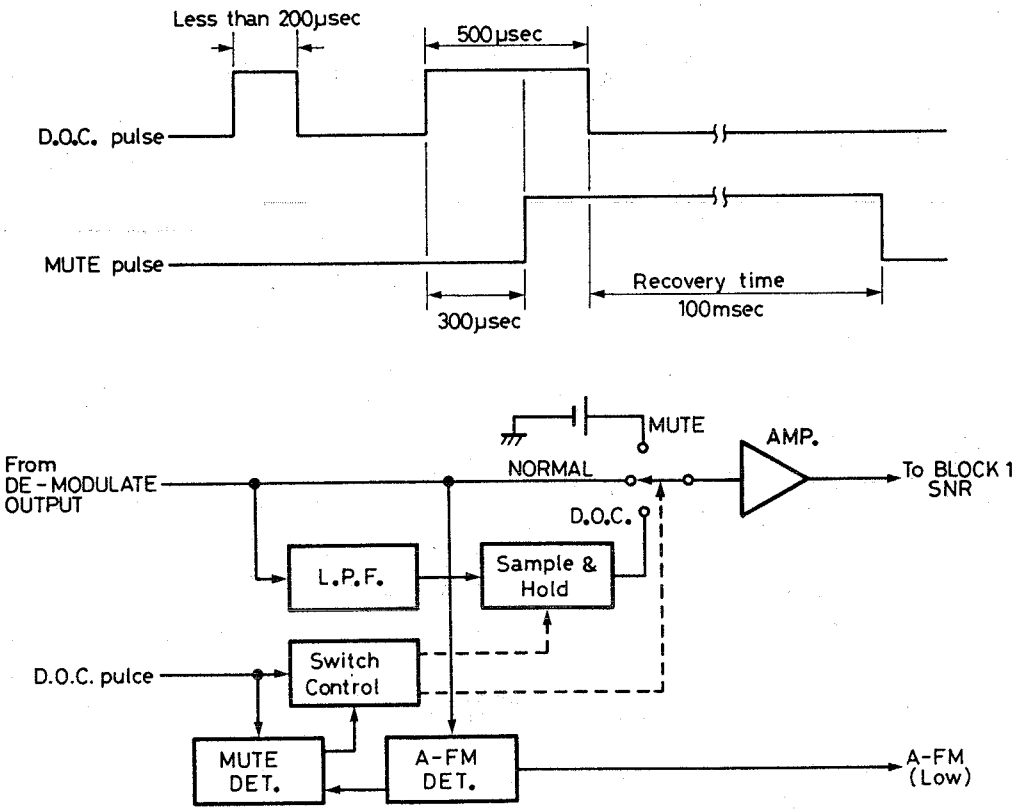


Fig. 2-4-22

3. AUDIO FM CIRCUIT

1. OUTLINE

As shown in Table 3-1-1, the ICs which make up the audio FM circuit in the VTC-M40 series and the VTC-M50 series are almost exactly the same. The circuit operation is also the same.

The audio FM circuit in the VTC-M40 series is composed of the AD-1 circuit board shown in Fig. 3-1-1. In addition to the audio FM circuit, the AD-1 circuit board also contains the normal audio circuit, the audio signal select circuit, and the chroma signal processing circuit for the video circuit system. Also, the preamplifier circuit for the special audio FM head is located on the AD-3 circuit board. The audio FM circuit mostly consists of new ICs developed for the PAL Beta hi-fi system, and most of the circuit operations are processed within the ICs. Thus the explanation of the audio FM circuit will be mainly an explanation of the important ICs, using the VTC-M40 as the basic model.

	IC Name	M40 Location No.	M50 Location No.	Function
AUDIO FM	TK15021	IC2201, IC2202	IC2301	SWITCHING
	NJM4558S	IC2203		OP AMP.
	M5278L05	IC2204, IC2205	IC2305, IC2306	REGULATOR
	CX20104	IC2208	IC2304	SSB CONVERTOR FM MODULATOR
	CX20105	IC2209, IC2210	IC2303, IC2403	AUDIO FM DE-MODULATOR
	CX20097	IC2211	IC2302	NOISE REDUCTION

Table 3-1-1

OVERALL WIRING DIAGRAM (AUDIO FM CIRCUIT)

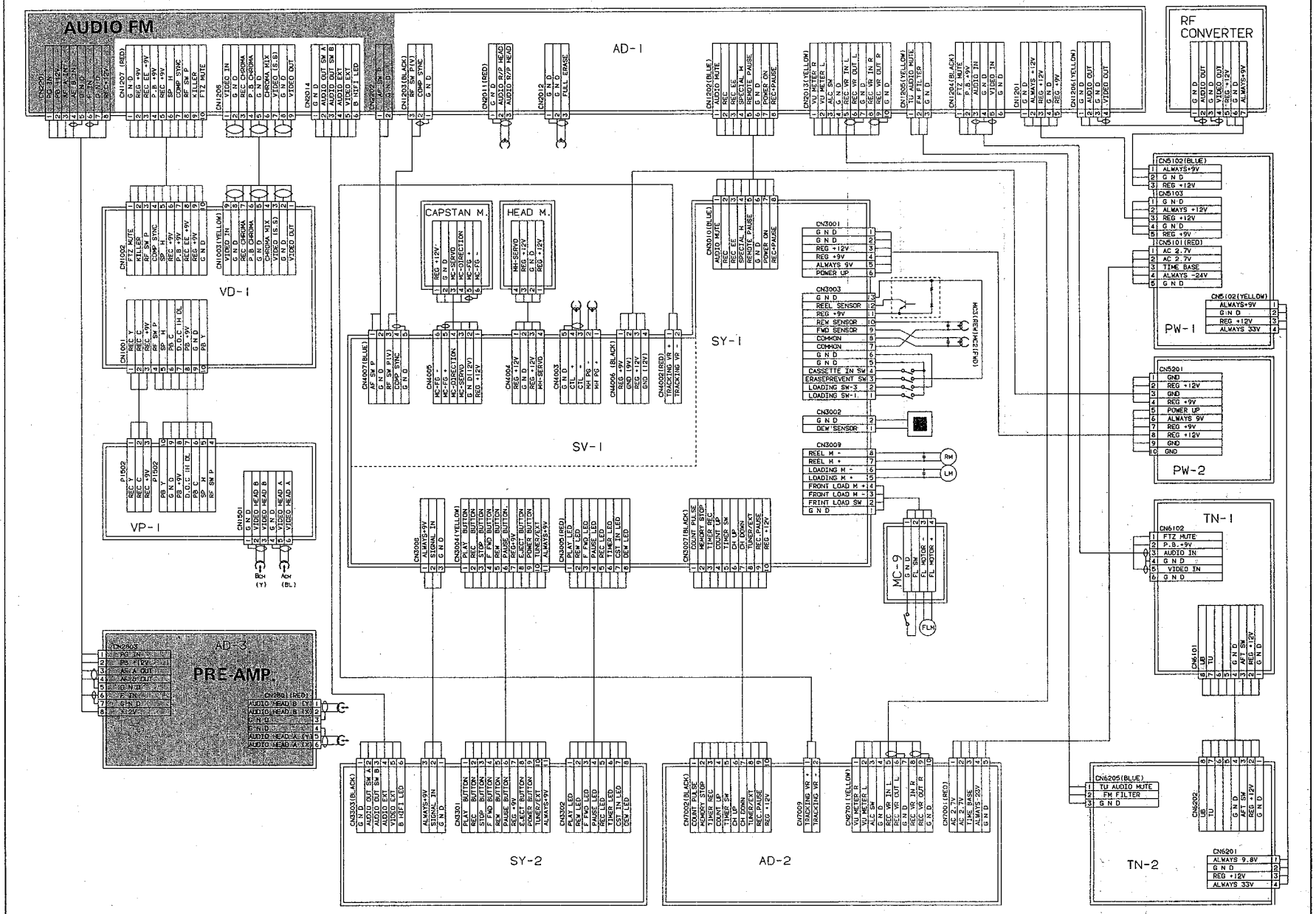


Fig. 3-1-1

2. AUDIO FM CIRCUIT COMPOSITION

The audio FM circuit is composed of five blocks: four bipolar ICs, CX20097, CX20104, and CX20105 (two of these are used), specially developed for use in the PAL Beta hi-fi system, and a recording/playback preamplifier. This block construction is the same as that shown in Fig. 2-4-1 in Chapter 2, Section 4, and the REC/PB AMP and BPFs (bandpass filters) shown in Fig. 3-2-1 correspond to the fifth block.

The ICs which make up the audio FM circuit in Fig. 3-2-1 are as follows:

1. Block 1: CX20097 (IC2211)
2. Block 2: CX20104 (IC2208)
3. Block 3: CX20105 (IC2209) left channel
4. Block 4: CX20105 (IC2210) right channel
5. Block 5: Transistor construction and two BPFs (F2205 and F2206)

AUDIO FM BLOCK DIAGRAM

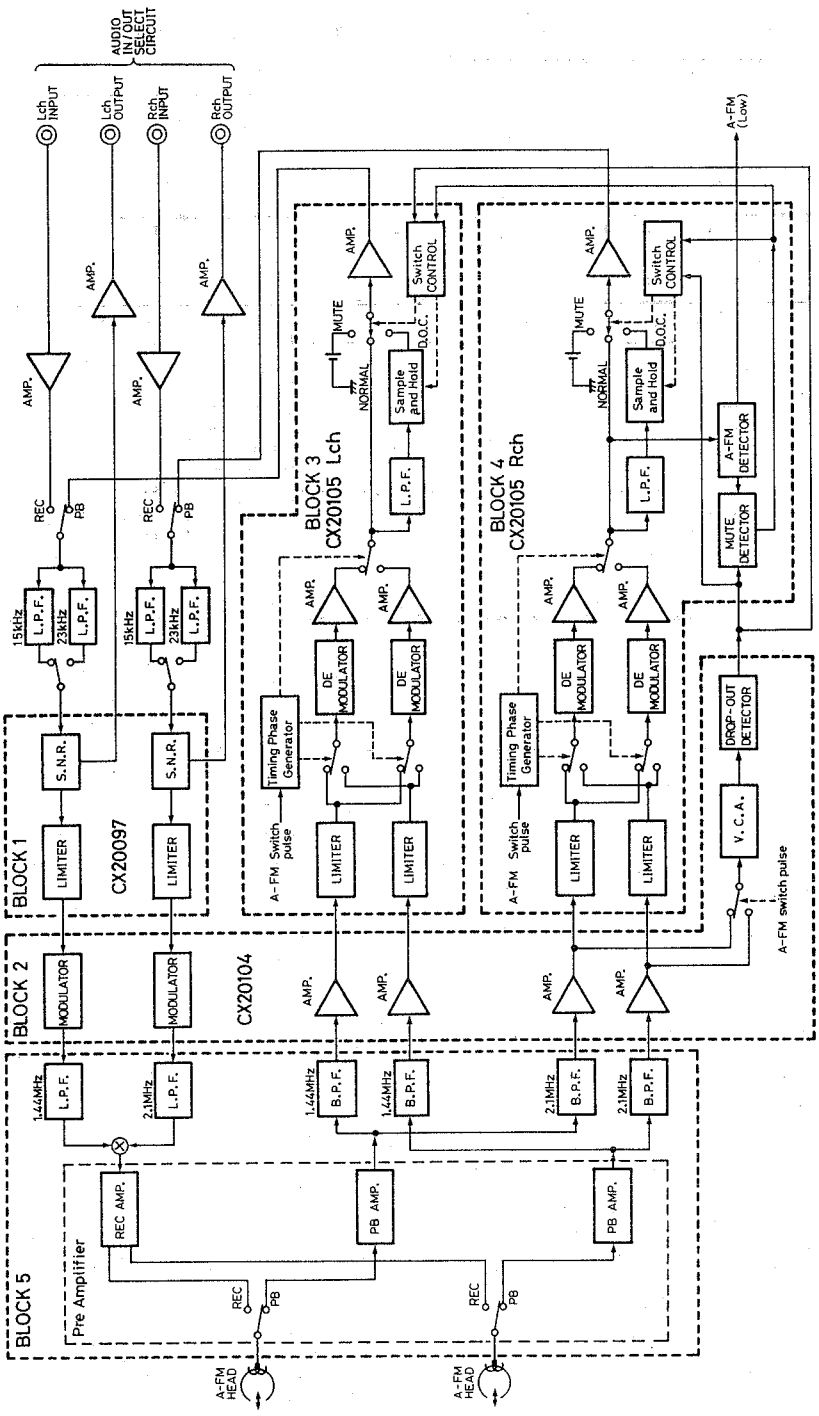


Fig. 3-2-1

The main functions of these ICs are explained next.

(1) BLOCK 1: CX20097

The function of CX20097 is SNR (silent noise reduction). Just as the noise reduction systems used in various audio tape recorders, this IC logarithmically compresses the audio signals during recording and expands them during playback, thus increasing the dynamic range.

In hi-fi VCRs, because the audio signal is frequency modulated, the triangular noise particular to frequency modulation must be reduced, and the switching noise caused by the rotating heads must be processed. Thus, on these points, the noise component is somewhat different from that found in audio tape recorders. For this reason, this IC is a special VCR noise reduction system designed with provisions for emphasis, etc., and it provides a dynamic range of 80 dB or more.

To perform this task, the following functions are included in the IC:

- VCA (voltage-controlled amplifier)
- Detector
- MOA (main operation amplifier)
- Adder
- Buffer amplifier
- Overmodulation limiter
- Overshoot limiter

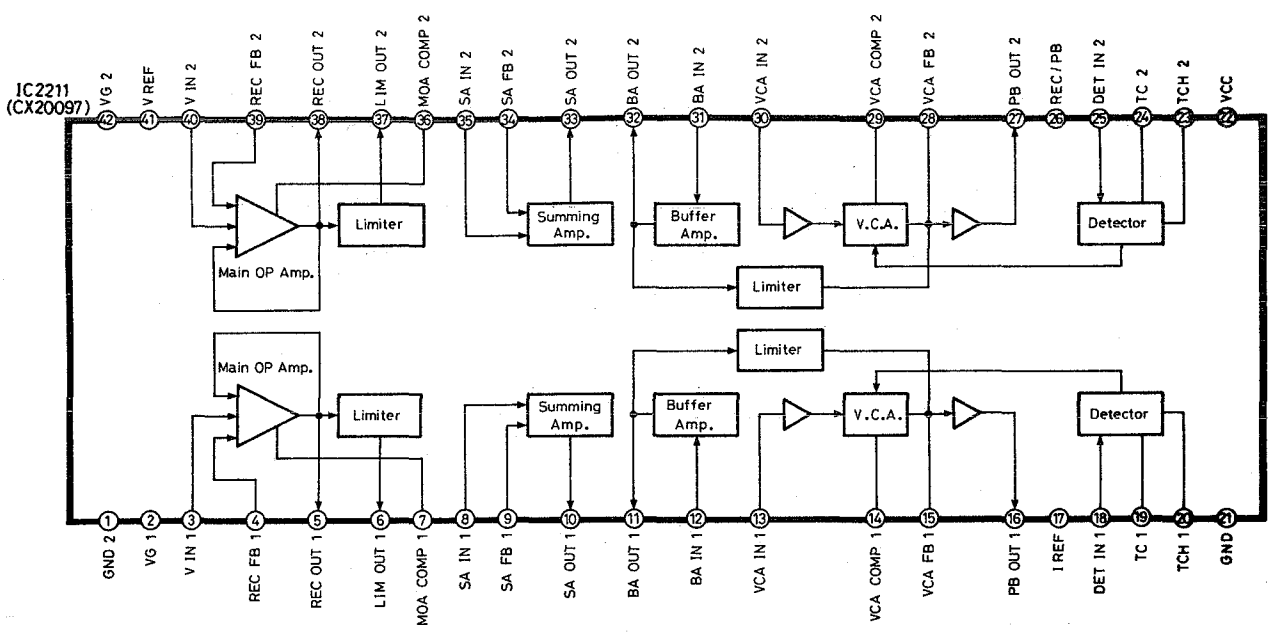


Fig. 3-2-2

[PIN FUNCTIONS]

Pin	Label	I/O	Terminal description
1	GND 2		GND of circuit which outputs VREF and VG (center point).
2	VG 1		Reference for channel 1 VG signal system.
3	V IN 1	I	Common recording/playback input terminal for channel 1.
4	REC FB 1	I	MOA inverting input terminal; feedback of channel 1 playback output during recording.
5	REC OUT 1	O	Channel 1 recording output.
6	LMT OUT 1	O	Channel 1 overmodulation limiter output.
7	MOA COMP 1		Channel 1 MOA phase compensation terminal.
8	SA IN 1	I	Non-inverting input terminal for channel 1 adder.
9	SA FB 1	I	Inverting input terminal for channel 1 adder.
10	SA OUT 1	O	Channel 1 adder output.
11	BA OUT 1	O	Channel 1 buffer amplifier output.
12	BA IN 1	I	Non-inverting input terminal for channel 1 buffer amplifier.
13	VCA IN 1	I	Channel 1 VCA input terminal. Current input and imaginary short for VG.
14	VCA COMP 1		Channel 1 VCA phase compensation terminal.
15	VCA FB 1	I	Inverting input terminal for channel 1 VCA output amplifier; performs I → V conversion.
16	PB OUT 1	O	Channel 1 playback output terminal.
17	IREF	I	Reference current input terminal for detectors of both channels.
18	DET IN 1	I	Channel 1 detector input terminal; current input.
19	TC 1		Smooths two-wave rectification waveform of channel 1 detector; also determines attack recovery time constant.
20	TCH 1		Determines hold time constant for channel 1 detector.
21	GND		GND for signal and control systems of both channels.

Pin	Label	I/O	Terminal description
22	VCC		Power supply terminal for both channels.
23	TCH 2		Determines hold time constant for channel 2 detector.
24	TC 2		Smooths two-wave rectification waveform of channel 2 detector; also determines attack recovery time constant.
25	DET IN 2	I	Channel 2 detector input terminal; current input.
26	REC/PB		REC/PB mode switching terminal.
27	PB OUT 2	O	Channel 2 playback output terminal.
28	VCA FB 2	I	Inverting input terminal for channel 2 VCA output amplifier; performs I → V conversion.
29	VCA COMP 2		Channel 2 VCA phase compensation terminal.
30	VCA IN 2	I	Channel 2 VCA input terminal. Current input and imaginary short to VG.
31	BA IN 2	I	Non-inverting input terminal for channel 2 buffer amplifier.
32	BA OUT 2	O	Channel 2 buffer amplifier output.
33	SA OUT 2	O	Channel 2 adder output.
34	SA FB 2	I	Inverting input terminal for channel 2 adder.
35	SA IN 2	I	Non-inverting input terminal for channel 2 adder.
36	MOA COMP 2		Channel 2 MOA phase compensation terminal.
37	LMT OUT 2	O	Channel 2 overmodulation limiter output.
38	REC OUT 2	O	Channel 2 recording output.
39	REC FB 2	I	MOA inverting input terminal; feedback of channel 2 playback output during recording.
40	V IN 2	I	Common recording/playback input terminal for channel 2.
41	VREF		Reference voltage for VCA gain.
42	VG 2		Reference for channel 2 VG signal system.

Table 3-2-1

The features of the IC are as follows:

- Built-in logarithmic compression/expansion compandor (compression ratio: 2:1)
- 30-dB dynamic range improvement
- Built-in overmodulation limiter and overshoot limiter

[Note]

Because the internal oscillation performed by the CX20097 IC used in the SNR circuit in the PAL Beta hi-fi system could possibly reduce the IC's performance, the IC has been replaced by CX20097A in the VTC-M50 series.

When this IC change is made, the peripheral circuits must also be changed as shown in Table 3-2-2. (Changing only the IC could result in incompatibility in the SNR circuit.)

Fig. 3-2-3 shows a screen of the parts in the peripheral circuits which must be changed when the IC is changed. This circuit diagram uses the left channel of the VTC-M40 series as a reference; the appropriate right channel parts and CR composite parts must also be changed. For further details, refer to the circuit diagrams of the individual models.

IC / SNR circuit	CX20097	CX20097A
CX20097 and peripheral circuit	———	Cannot be used
CX20097A and peripheral circuit	Can be used	———

Table 3-2-2

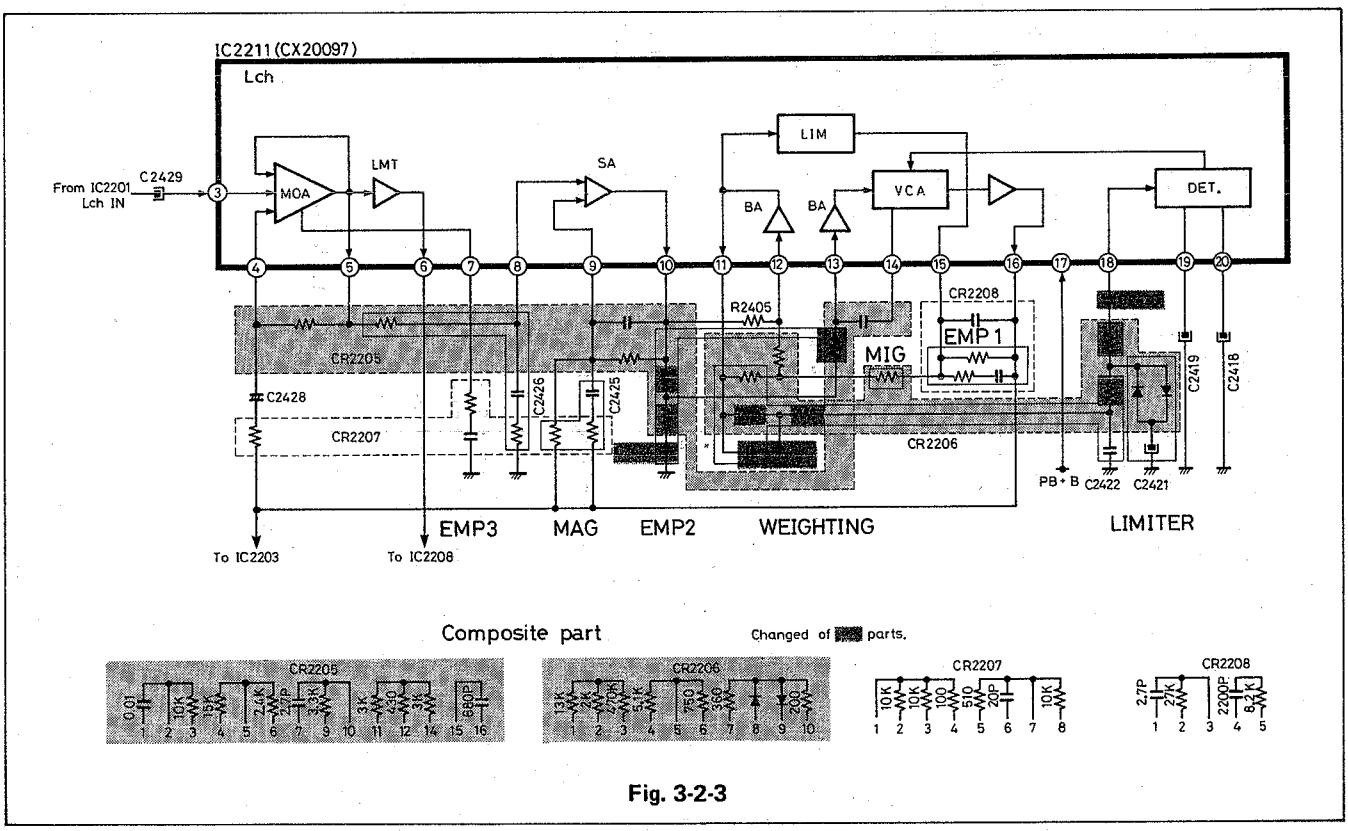


Fig. 3-2-3

(2) BLOCK 2: CX20104

CX20104 is a bipolar IC developed for use in processing the audio FM signal in Beta hi-fi VCRs. CX20104 performs the following operations during recording and playback.

[During recording]

- The IC functions as a frequency modulator and frequency modulates the compressed audio signal from the SNR circuit with good linearity.
- Frequency modulation is done at the maximum deviation (±200 kHz) with respect to the carrier frequencies (1.44 MHz for the left channel and 2.1 MHz for the right channel) of the audio signals.

[During playback]

- The IC inputs the preamplifier circuit output signal to the demodulator, where it is amplified to the rated input level.
- The IC detects tape drop-out and outputs the control signal (drop-out pulse) for the various compensation circuits.
- The drop-out pulse is used by CX20105 in the pre-hold circuit and mute detection circuit.

In order to perform these operations, the IC is equipped with the following functions:

- Frequency modulator
- SSB converter
- Recording AGC circuit
- Playback AGC circuit
- Drop-out detection circuit
- 150-kHz oscillator

In addition, the features of the IC are as follows:

- Built-in modulator with low distortion rate (0.3% or less)
- Built-in SSB converter for use in both recording and playback
- Recording level of four carriers can be adjusted at one location
- four built-in playback amplifier circuits
- Built-in VCA for control of playback RF level to specified level
- Performs drop-out detection using absolute value (detection level can be adjusted externally)
- Built-in 150-kHz Wien-bridge oscillator (low distortion rate)

(Used only in the NTSC Beta hi-fi format, not in the PAL Beta hi-fi format.)

Note-

Table 3-2-3 is a general function description of the IC input/output pins. This table corresponds to the functions of the IC itself in the PAL and NTSC formats; however, when the IC is actually used and assembled into the circuitry, the functions and uses may change somewhat according to the features of the particular model and the relationship with peripheral circuits.

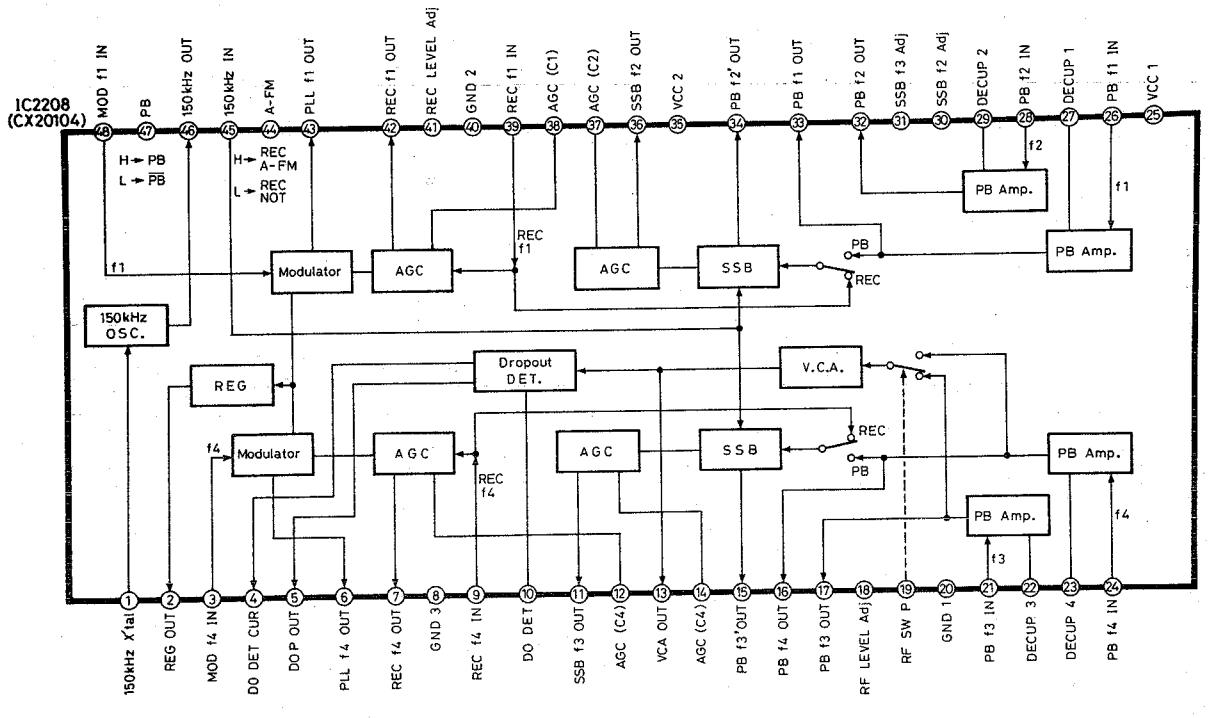


Fig. 3-2-4

[PIN FUNCTION]

Pin	Label	I/O	Terminal description
1	150kHz X'tal		Crystal connection terminal; P3 type 149.47-kHz crystal is directly connected for oscillation.
2	REG OUT	O	REG output terminal (VREG = 4.2 V) for power supply ripple suppression; this output is used to drive amplifier.
3	MOD f_4 IN	I	Terminal for input of current which determines MOD 2 oscillation frequency; frequency modulation is achieved by inputting audio signal via resistance.
4	D.O. DET. CUR		Dropout detection current is determined by externally connected resistance.
5	D.O. P OUT	O	Dropout detection pulse output terminal During dropout: 3.3 V (typical) Normally: 0 V (typical)
6	PLL f_4 OUT	O	MOD 2 PLL output terminal.
7	REC f_4 OUT	O	MOD 2 REC f_4 output terminal; converted to a sine wave by low-pass filter and then used as AGC input and REC AMP input.
8	GND 3		GND terminal for REG and MOD.
9	REC f_4 IN	I	f_4 input terminal for AGC and SSB 2.
10	D.O. DET		Terminal which determines dropout detection level; −12 dB at 1.0 V (variable range: −5 to −18 dB).
11	SSB f_3 OUT	O	REC f_3 OUT terminal; interference: −25 dB or less.
12	AGC (C_4)		Capacitor connection terminal for REC f_4 AGC loop filter.
13	VCA OUT	O	Dropout level adjustment terminal for adjusting differences among individual VCRs; play reference tape and adjust pin 18 voltage to obtain output of 1.9 Vp-p.
14	AGC (C_3)		Capacitor connection terminal for REC f_3 AGC loop filter.
15	PB f_3' OUT	O	PB SSB f_3' output terminal.
16	PB f_4 OUT	O	PB f_4 AMP output terminal (gain: 40 dB).
17	PB f_3 OUT	O	PB f_3 AMP output terminal (gain: 40 dB).
18	RF LEVEL Adj		VCA adjustment terminal.
19	RF SW P	I	RF SW P input terminal; switches output of PB f_3 AMP and PB f_4 AMP.
20	GND 1		PB GND terminal.
21	PB f_3 IN	I	PB f_3 AMP input terminal.
22	DECUP 3		PB f_3 AMP decoupling terminal.
23	DECUP 4		PB f_4 AMP decoupling terminal.
24	PB f_4 IN	I	PB f_4 AMP input terminal.
25	VCC 1		PB power supply terminal.
26	PB f_1 IN	I	PB f_1 AMP input terminal.
27	DECUP 1		PB f_1 AMP decoupling terminal.
28	PB f_2 IN	I	PB f_2 AMP input terminal.
29	DECUP 2		PB f_2 AMP decoupling terminal.
30	SSB f_2 Adj		Adjustment terminal for unneeded sideband leak of SSB 1 output.
31	SSB f_3 Adj		Adjustment terminal for unneeded sideband leak of SSB 2 output.
32	PB f_2 OUT	O	PB f_2 AMP output terminal (gain: 40 dB).
33	PB f_1 OUT	O	PB f_1 AMP output terminal (gain: 40 dB).
34	PB f_2' OUT	O	PB SSB f_2' output terminal.
35	VCC 2		Power supply terminal for SSB, REC AGC, and 150-kHz OSC.
36	SSB f_2 OUT	O	REC f_2 OUT terminal; interference: −25 dB or less.
37	AGC (C_2)		Capacitor connection terminal for REC f_2 AGC loop filter.
38	AGC (C_1)		Capacitor connection terminal for REC f_1 AGC loop filter.
39	REC f_1 IN	I	f_1 input terminal for AGC and SSB 1.
40	GND 2		GND terminal for SSB, REC AGC, and 150-kHz OSC.
41	REC LEVEL Adj.		REC level adjustment terminal: open (0.95 V) → REC output 0.86 Vp-p 0.26 V → −3 dB; 1.94 V → +3 dB
42	REC f_1 OUT	O	MOD 1 REC f_1 output terminal; converted to sine wave by low-pass filter and then used as AGC input and REC AMP input.
43	PLL f_1 OUT	O	MOD 1 PLL output terminal.
44	AFM		AFM mode switch terminal: HIGH (min. 4.0 V) → AFM REC mode LOW (max. 1.0 V) → not REC mode
45	150kHz IN	I	SSB 150-kHz input terminal.
46	150kHz OUT	O	150-kHz OSC output terminal.
47	PB		PB mode switch terminal: HIGH (min. 4.0 V) → PB mode LOW (max. 1.0 V) → $\overline{\text{PB}}$ mode
48	MOD f_1 IN	I	Terminal for input of current which determines MOD 1 oscillation frequency; frequency modulation is achieved by inputting audio signal via resistance.

Table 3-2-3

(3) BLOCKS 3 and 4: CX20105

CX20105 is a bipolar IC developed for use in processing the various signals needed in the frequency demodulation of Beta hi-fi VCRs (NTSC and PAL).

CX20105 performs the following operations:

- Modulation free of switching noise using two demodulators per channel
- Pre-hold operation when drop-out occurs
- Mute circuit compensation
- Identification of audio FM mode or normal audio mode.

In order to perform these operations, the IC is equipped with the following functions:

- Limiter
- Frequency demodulators
- Timing pulse signal generator
- Switch for RF switching noise processing
- Drop-out compensation circuit
- Beta hi-fi discrimination circuit
- Mute signal detection circuit
- Mute switch
- Amplifier for demodulated signal
- Buffer amplifier

In addition, the IC also has the following features:

- Built-in limiter reduces the effects of power supply ripple.
- Low distortion rate
- Frequency response of 20 kHz or more
- Two demodulators per audio track as used in the RF signal switching noise processing system, and the switching timing for the demodulators is staggered in order to provide noise-free demodulator output.
- Identification of audio FM mode or normal audio mode
- Built-in drop-out compensation and mute circuits

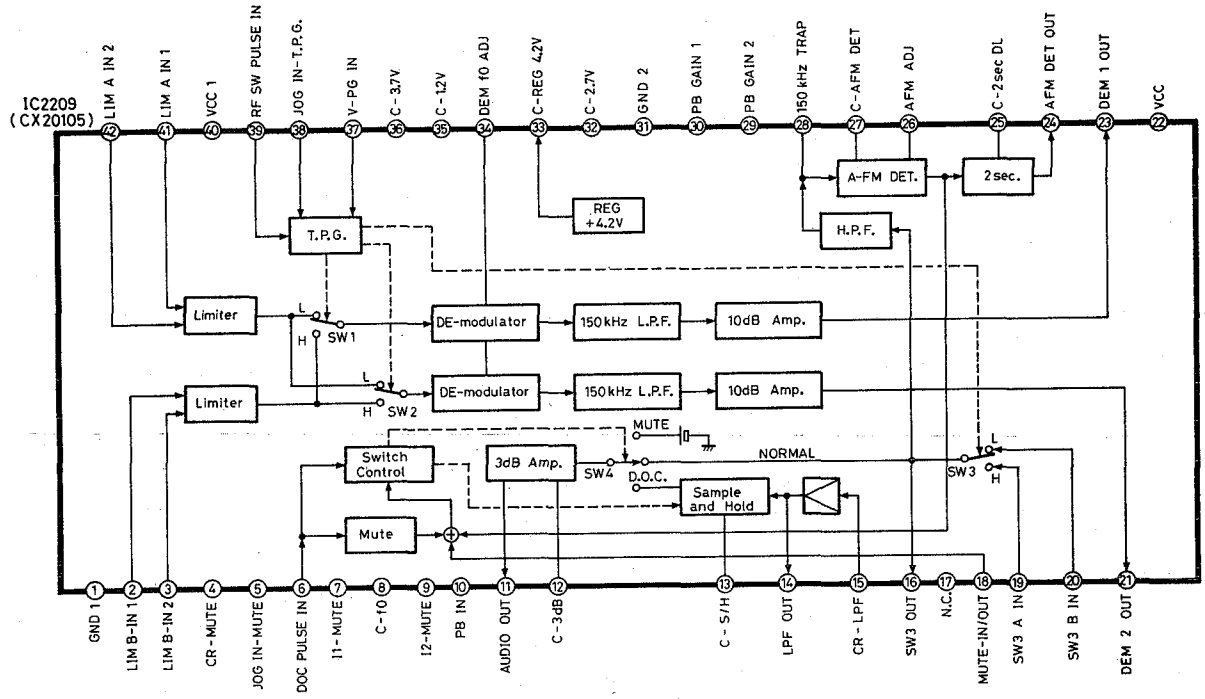


Fig. 3-2-5

[PIN FUNCTION]

Pin	Label	I/O	Terminal description
1	GND 1		GND for limiter, demodulators, and TPG.
2	LIM B IN 1	I	Limiter differential input pin.
3	LIM B IN 2	I	Limiter differential input pin.
4	CR-MUTE		Time constant connection pin for dropout compensator mute signal detection.
5	JOG IN-MUTE		Because dropout pulse input to pin 6 increases during JOG mode, this terminal increases, pin 6 input threshold at which dropout compensator mute functions. (During JOG mode, pin 5 is HIGH; during other modes it is LOW.)
6	D.O. P IN	I	Dropout pulse input pin.
7	I_1-MUTE		Determines AFM detection delay time and charge current for dropout compensator mute signal detection.
8	C-f_0		Decoupling of current source of circuit which determines DEM-f_0 (demodulator's intermediate frequency).
9	I_2-MUTE		Determines discharge current for dropout compensator mute signal detection.
10	PB IN	I	Playback input pin. (When PB is LOW, IC is in current-saving mode and pin 24 (AFM DET OUT) is forced to become LOW.)
11	AUDIO OUT	O	Playback audio signal output pin.
12	C-3dB		3-dB amplifier decoupling. (When open, amplifier gain becomes 0 dB.)
13	C-S/H		Capacitor connection pin for sample and hold.
14	LPF OUT	O	Sample and hold pre-stage low-pass filter output pin (CR connection).
15	CR-LPF		CR connection pin for low-pass filter mentioned above.
16	SW3 OUT	O	Input pin for low-pass filter mentioned above (CR connection).
17	N.C.		Not connected.
18	MUTE IN/OUT		Dropout compensator mute output (HIGH during mute and LOW at other times); also input of system mute signal (externally forced to become HIGH).
19	SW3 A IN	I	Inputs pin 23 output to SW3 (coupling capacitor connected between pins 19 and 23).
20	SW3 B IN	I	Inputs pin 21 output to SW3 (coupling capacitor connected between pins 20 and 21).
21	DEM 2 OUT	O	Outputs signal amplified 10 dB from demodulator 2 via 150-kHz low-pass filter.
22	Vcc 2		Circuit's 5-V power supply (power supply for everything except limiter, demodulators and TPG).
23	DEM 1 OUT	O	Outputs signal amplified 10 dB from demodulator 1 via 150-kHz low-pass filter.
24	AFM DET. OUT	O	AFM discrimination output pin (HIGH during AFM playback (including CUE and REVIEW) and LOW at all other times).
25	C-2 sec DL		Connection of time constant (capacitor) for approx. 2-sec delay of AFM discrimination output.
26	AFM Adj		Adjustment of AFM discrimination threshold level (can be open when used in NTSC format).
27	C-AFM DET		Capacitor connection pin for integration of noise rectified in AFM discrimination circuit.
28	150kHz TRAP		LC trap connection pin for removal of 150-kHz component in AFM discrimination circuit input signal (NTSC only).
29	PB GAIN 2		Demodulation level adjustment pin for demodulator 2 (channel B).
30	PB GAIN 1		Demodulation level adjustment pin for demodulator 1 (channel A).
31	GND 2		GND for everything except limiter, demodulators, and TPG.
32	C-2.7V		Audio switch and buffer decoupling terminal.
33	C-REG 4.2V		Regulator 4.2-V output pin (for decoupling).
34	DEM f_0 Adj		Pin for external adjustment of f_0 of demodulators 4 and 2. (Adjust I_0 so that pin 34 = 2.7 V (constant) and pin 21 = 2.00 V.)
35	C-1.2V		Demodulator reference voltage (for decoupling).
36	C-3.7V		Demodulator reference voltage (for decoupling).
37	V-PG IN	I	Video RA PG input pin; used in 3-head (A, B, A') VCRs. When pin 38 (JOG) = HIGH, SW1 = VPG, SW2 = HIGH, and SW3 = HIGH.
38	JOG IN-TPG	I	JOG input pin; in 3-head VCR, HIGH during JOG mode.
39	RF SW PULSE IN	I	RF switching pulse input pin. When pin 38 (JOG) = LOW, SW1, 2, 3 = HIGH for HIGH RF switching pulse and LOW for LOW RF switching pulse.
40	Vcc 1		Circuit's 5-V power supply (for limiter, demodulators, and TPG).
41	LIM A IN 1	I	Limiter differential input pin.
42	LIM A IN 2	I	Limiter differential input pin.

Table 3-2-4

(4) BLOCK5: PLAYBACK PREAMPLIFIER AND BAND-PASS FILTERS (BPF)

This circuit includes a power amplifier for supplying the recording current for the heads to record onto the tape the audio FM signals output from the demodulators and a playback amplifier for preamplification of the output from the heads during playback. In order to perform amplification with a good S/N ratio, it also has a step-up transformer and high-gm (mutual conductance, just as the h_{fe} of a transistor) FET inputs just like those of the video amplifier.

The BPFs are LC composite components for separation of the 1.44-MHz (left channel) and 2.10-MHz (right channel) components from the audio FM signal.

Fig. 3-2-6 shows the characteristics of the 1.44-MHz BPF and Fig. 3-2-7 shows the characteristics of the 2.10-MHz BPF.

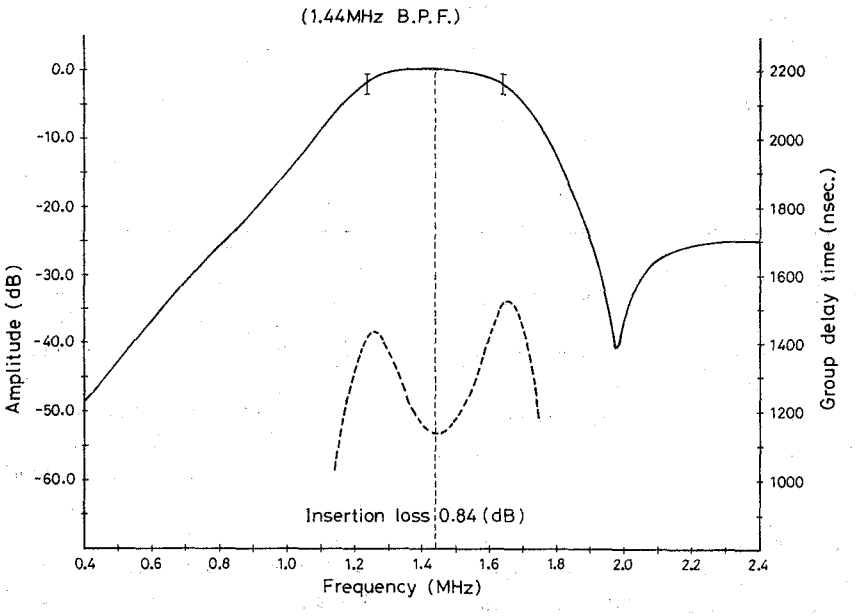


Fig. 3-2-6

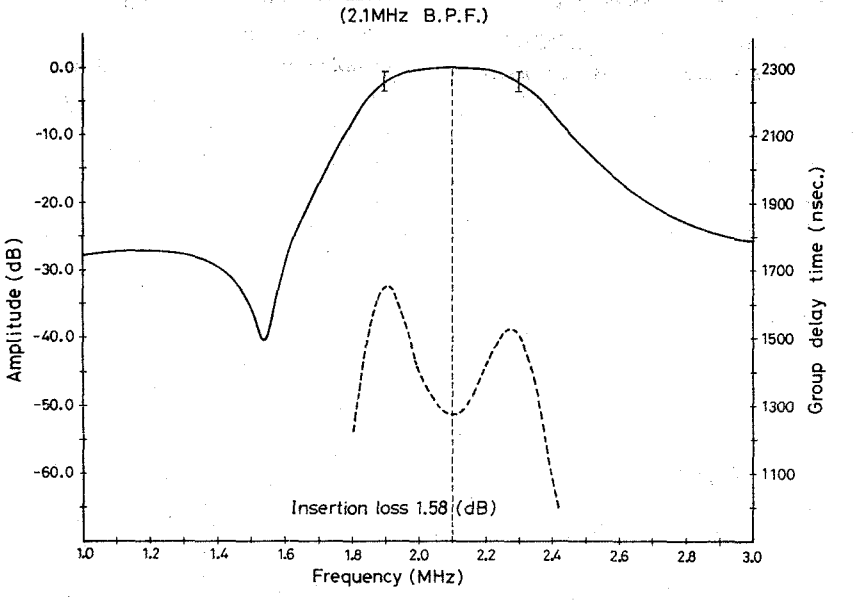


Fig. 3-2-7

3. DESCRIPTIONS OF THE MAIN AUDIO FM BLOCKS

The following explanation deals primarily with the functions of the main ICs which make up the audio FM circuit, devided into three large blocks.

1. SNR block: CX20097 (IC2211)
2. Frequency modulation block: CX20104 (IC2208)
3. Audio frequency demodulation block: CX20105 (IC2209 and IC2210)

(1) SNR BLOCK

A. REC mode operation description

During the REC mode, the SNR block compresses the audio signals input to the audio FM circuit and then generates the audio signals which are frequency modulated by the FM block and deep-layer recorded onto the tape by the A-FM heads. Also, for the E-E monitor, the audio signals not compressed by the SNR block are extracted and output to the audio signal select circuit. The audio signals which are input to the SNR block during recording have been selected by a switch in IC2201 by passing them through a 15-kHz filter and a 23-kHz filter.

This control is done by the REC FM LOW signal. During recording, if the tuner or line input is an FM signal, pins 7 and 8 of IC2201 are connected; if it is not an FM signal, pins 9 and 8 of IC2201 are connected. Also, during playback, the connection is fixed to pins 9 and 8 of this same IC.

The noise and carrier components of the left-channel audio signal input to the SNR block above 20 kHz are removed by the F2202 23-kHz filter. Also, the F2204 15-kHz filter attenuates the leakage signal of the 19-kHz pilot signal contained in the television's FM tuner input and line input signals, and the fH (15.625 kHz) signal from the VCR's FM tuner.

The audio signal output from pin 8 of IC2201 is then input to the SNR circuit centering around IC2211 (CX20097).

After the dynamic range of the audio signal is compressed by the SNR circuit, it is output from pin 6 of IC2211 to IC2208 (CX20104), which is the frequency modulator.

In the same way, after the audio signal is amplified approximately 16.5 dB by the IC2203 operational amplifier (RF LINE AMP) as the audio monitor, it is output from pin 4 of IC2211 to the audio signal select circuit.

The amount of amplification in the RF LINE AMP can be calculated as follows:

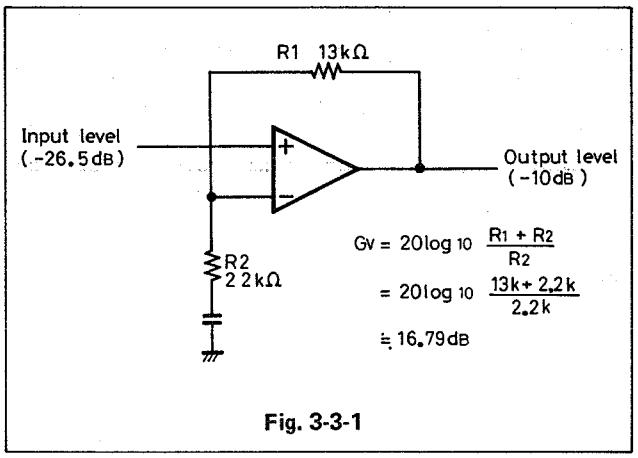


Fig. 3-3-1

The amplification of the RF LINE AMP is set to a higher level (16.79 dB) than the difference (16.5 dB) between the input/output levels in order to compensate the gain which occurs when the signals pass through the audio signal select circuit and other circuits so that an output level of −10dB can be obtained.

B. PB mode operation description

During the PB mode, the SNR block first expands the playback audio signals demodulated by the audio frequency demodulation block and then outputs them to the audio signal select circuit.

For the left channel audio signal input to the SNR block during playback, after the audio FM signal is demodulated by IC2209 and passed through the IC2202 REC/PB switch, the noise and carrier components above 20 kHz are removed by the F2202 23-kHz filter. Also, the audio signal output from pin 8 of IC2208 is expanded by the SNR circuit centering around IC2211 (CX20097) just as during recording, and then the signals are output from pin 4 of IC2211 to the audio signal select circuit. At this time, just as during recording, the audio signals are output to the audio signal select circuit after being amplified approximately 16.5 dB by IC2003.

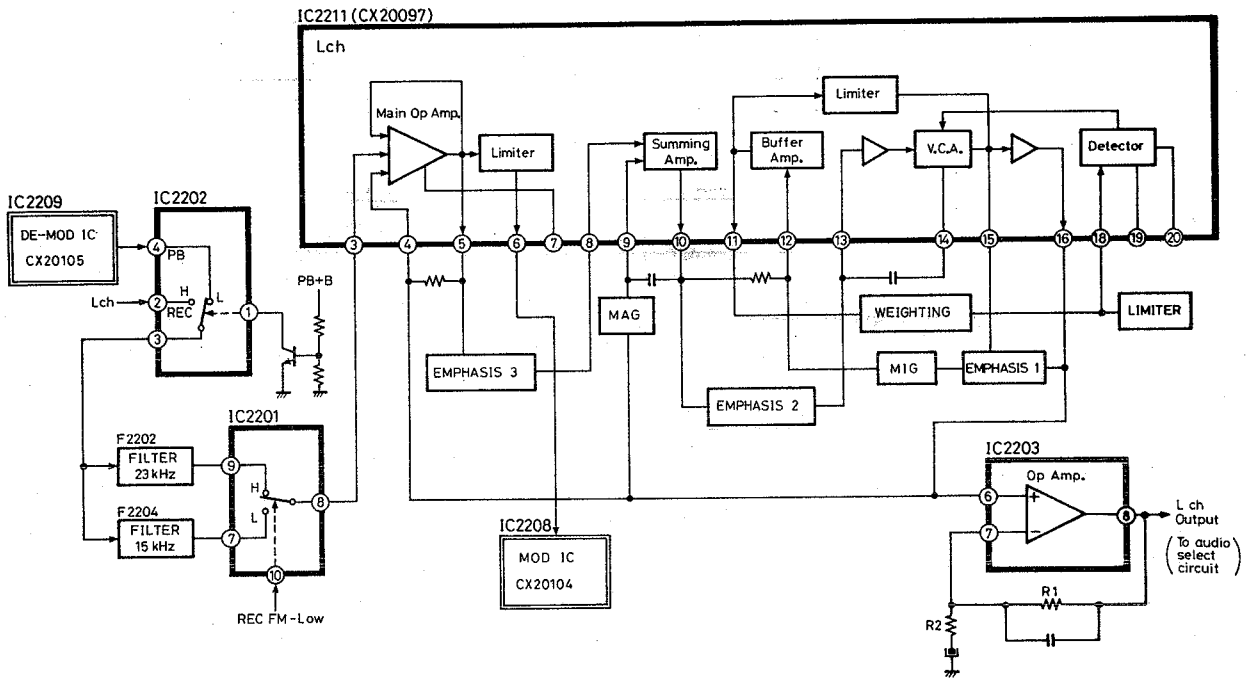


Fig. 3-3-2

C. Silent noise reduction circuit

The SNR block diagram of the SNR circuit in the PAL Beta hi-fi system is shown in Fig. 3-3-3, and the SNR circuit composition for the left channel is shown in Fig. 3-3-4. The calculated values for the noise reduction of the SNR circuit are as follows:

100 Hz: 30.0 dB
1 kHz: 30.5 dB
10 kHz: 41.5 dB

In addition, the main characteristics of the Beta hi-fi system's dynamic emphasis are as shown here.

● Timer Constant

(Reference values)

EMPHASIS – 1	77.4μs • 18.0 μs	2.06 kHz • 8.82 kHz
EMPHASIS – 2	158 μs • 35.3 μs	1.01 kHz • 4.51 kHz
EMPHASIS – 3	186 μs • 150 μs	0.856 kHz • 1.06 kHz
	187 μs • 150 μs	0.850 kHz • 1.06 kHz
MAG PASS	300 μs • 150 μs	0.531 kHz • 1.06 kHz
WEIGHTING	1.34ms • 158 μs	119 Hz • 1.00 kHz
	12.2μs • 39.0 μs	13.0 kHz • 4.08 kHz

Table 3-3-1

● Frequency response

Encode mode (Allowable deviation: ± 2dB)

	50 Hz	100 Hz	200 Hz	400 Hz	700 Hz	1 kHz	2 kHz	4 kHz	7 kHz	10 kHz	15 kHz
–10	– 7.3	– 9.1	–10.0	–10.0	– 9.6	– 9.1	– 3.2	– 6.3			
–20	–13.2	–15.1	–16.1	–16.2	–16.0	–15.8	–15.3	–13.9	–12.0	–10.5	
–30	–18.9	–20.8	–21.8	–22.1	–21.9	–21.6	–21.0	–19.4	–17.6	–16.1	–14.3
–40	–24.4	–26.3	–27.5	–27.8	–27.6	–27.2	–26.5	–24.6	–22.8	–21.5	–19.7
–50	–30.1	–31.9	–32.9	–33.2	–33.0	–32.7	–31.7	–29.9	–28.1	–26.8	–25.1

Table 3-3-2 (dBs)

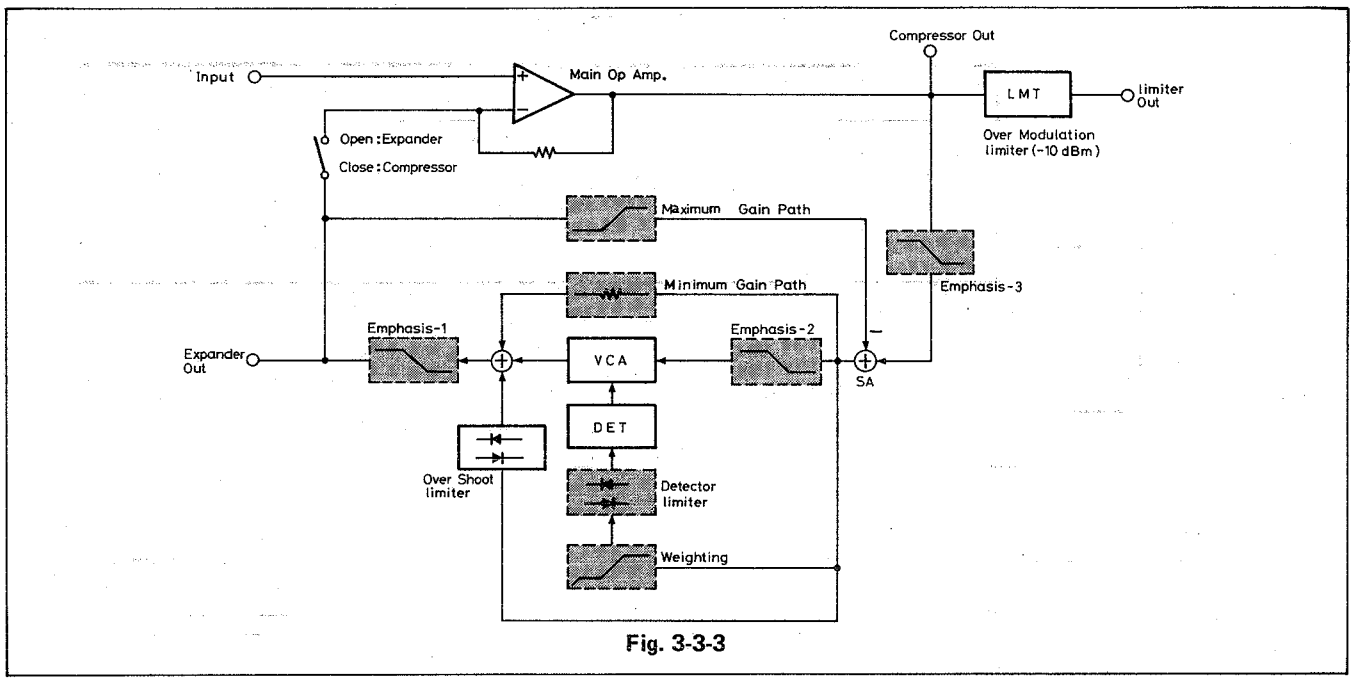


Fig. 3-3-3

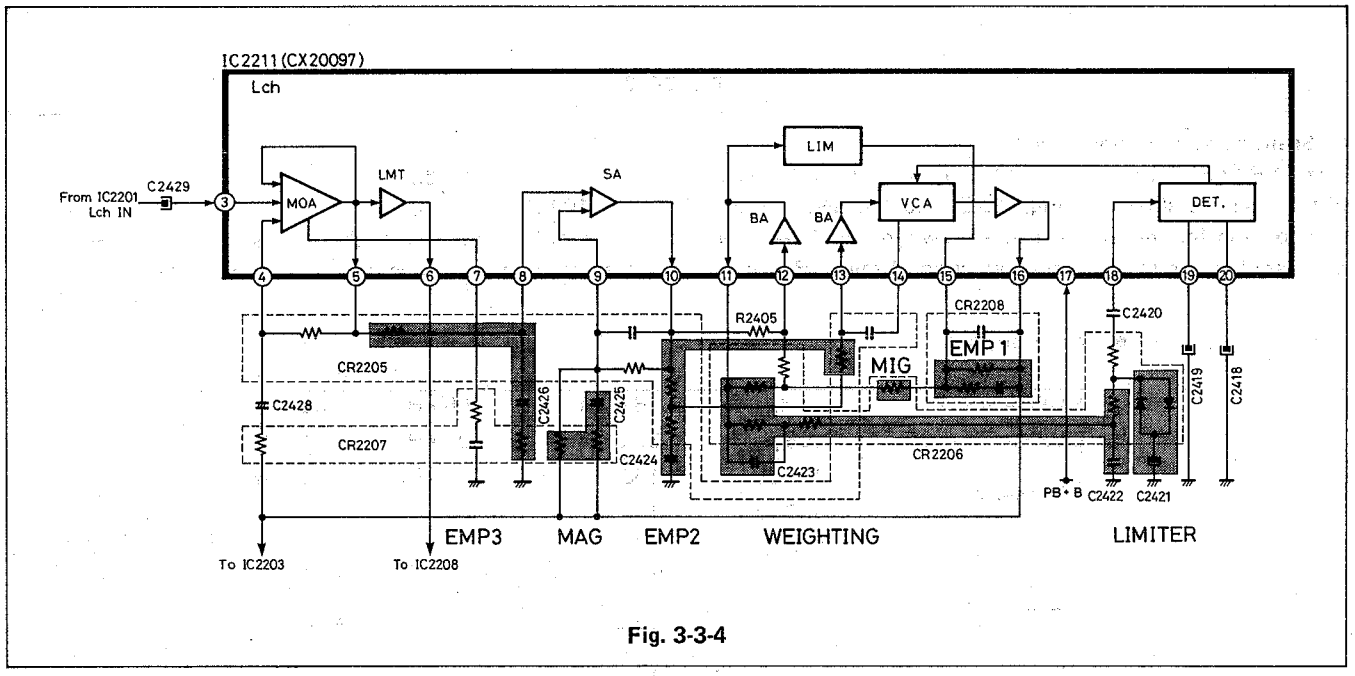


Fig. 3-3-4

D. Functions and characteristics of the SNR block

(1) MOA (main operational amplifier)

An internal switch allows the MOA to function as a voltage follower during playback; during recording the encode characteristic is obtained by including the decoder circuit in the MOA's negative-feedback loop. The MOA's unity gain is determined by the phase compensation time constant (pins 7 and 36).

(2) LMT (limiter amplifier)

The LMT is an overmodulation limiter. The REC OUT output becomes the LMT's input, and the limiting level is fixed at −10 dBm. For signals below −10 dBm, the LMT functions as a voltage follower.

(3) SA (summing amplifier)

When the input level is large, the SA is used to apply negative feedback from the playback output with respect to the playback input.

(4) BA (buffer amplifier)

The BA drives the detector and the limiter (LIM) which processes the overshoot.

(5) VCA (voltage-controlled amplifier)

The VCA is composed of a current input/current output current divider. Impedance elements (combined with emphasis) for voltage-current and current-voltage conversion are connected before and after the VCA.

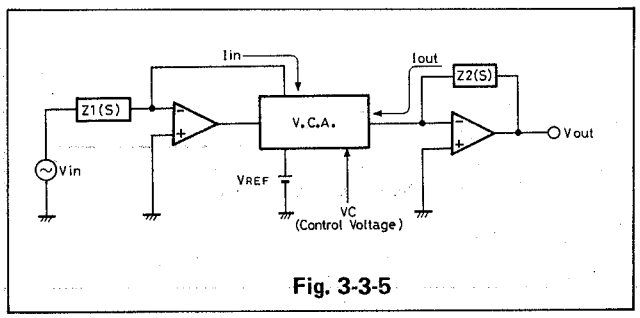


Fig. 3-3-5

The VCA's control sensitivity is 0.33 dB/mV, and the VCA gain is determined by the V_{REF}. Thus, when (V_{REF}−V_C) = 30 mV, the VCA gain is 10 dB.

If the VCA gain in the preceding figure is G,

$$V_{OUT} = V_{IN} \cdot G \cdot Z_2\,(S)/Z_1\,(S)$$

With this VCA, if $I_O = I_{IN} + I_{OUT}$, the maximum value of I_O depends on the power supply voltage. In other words,

V_{CC} = 5.0V → I_{omax} ≅ 1.7mAp-p

V_{CC} = 4.2V → I_{omax} ≅ 1.4mAp-p

(6) DET (detector)

In order for the detector to be able to accommodate a wide dynamic range, the current is input and the detector performs logarithmic conversion.

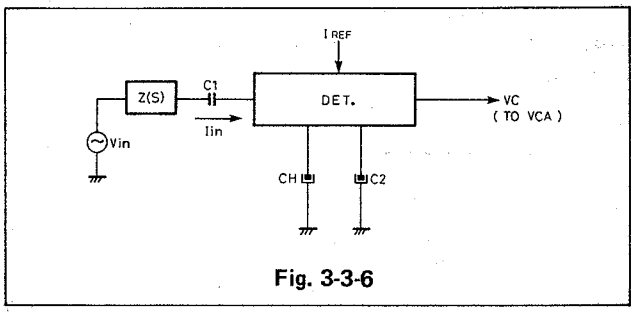


Fig. 3-3-6

Because the detector input is DC biased, it must be DC cut by C1. The input current I_{IN} at this time is:

$$I_{IN} = V_{IN}/Z\,(S)$$

The VCA's control voltage V_C is a function of the ratio of I_{IN} to I_{REF}. When I_{REF} = 100 μA, I_{IN} = 81.0 μArms and $V_C = V_{REF}$ (VCA gain = 0 dB).

The recovery time constant is determined by C2. Because it is relatively short, the ripple component in the detector output increases and the low-range distortion caused by cross modulation also increases. Thus the peak of the two-wave rectification waveform is held by CH in order to improve the distortion factor at a short recovery time constant.

(7) EMP-1 (emphasis 1)

The equation for calculating the time constant for EMP-1 is as follows:

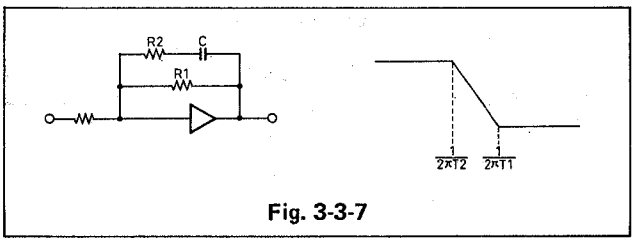


Fig. 3-3-7

$$R1//(R2 + \frac{1}{SC})$$

$$\frac{R1(R2 + \frac{1}{SC})}{R1 + R2 + \frac{1}{SC}} = \frac{R1(1 + SCR2)}{1 + SC(R1 + R2)} = R1 \cdot \frac{1 + ST1}{1 + ST2}$$

T1 = CR2 ≈ 18.0μsec (8.82kHz)

T2 = C (R1 + R2) ≈ 77.4μsec (2.06kHz)

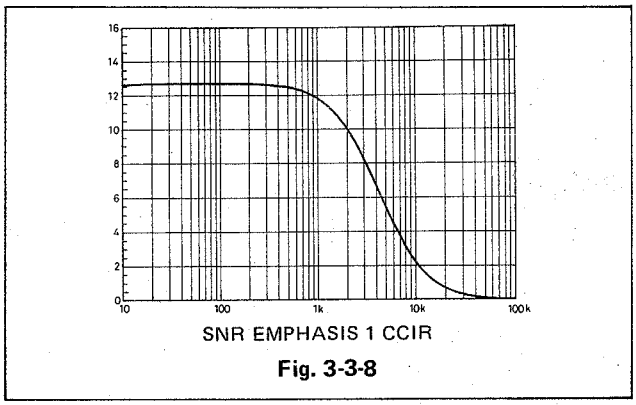


Fig. 3-3-8

(8) EMP-2 (emphasis 2)

The equation for calculating the time constant for EMP-2 is as follows:

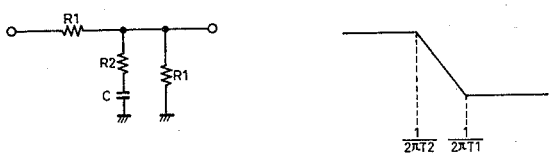
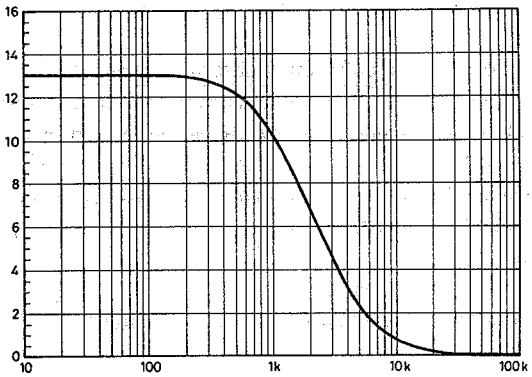


Fig. 3-3-9

$$\frac{1}{2} \cdot \frac{1 + SCR2}{1 + SC(R1/2 + R2)} = \frac{1}{2} \cdot \frac{1 + ST1}{1 + ST2}$$

$T1 = CR2 \approx 35.3\mu sec$ (4.51kHz)

$T2 = C\,(R1/2) \approx 158\mu sec$ (1.01kHz)



SNR EMPHASIS 2 CCIR

Fig. 3-3-10

(9) EMP-3 (emphasis 3)

The equation for calculating the time constant for EMP-3 is as follows:

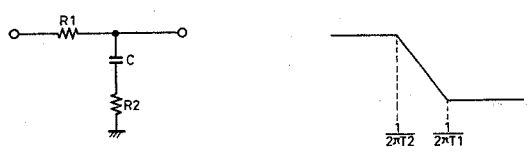
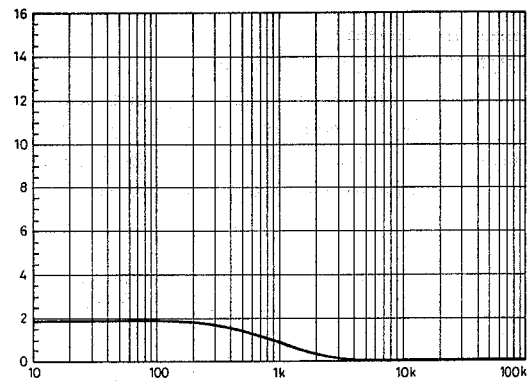


Fig. 3-3-11

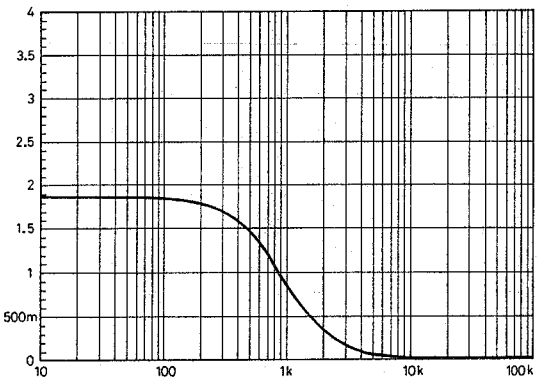
$$\frac{R2 + \frac{1}{SC}}{R1 + R2 + \frac{1}{SC}} = \frac{1 + SCR2}{1 + SC(R1 + R2)} = \frac{1 + ST1}{1 + ST2}$$

$T1 = CR2 \approx 150\mu sec$ (1.06kHz)

$T2 = C\,(R1 + R2) \approx 186\mu sec$ (0.856kHz)



SNR EMPHASIS 3 CCIR



SNR EMPHASIS 3 CCIR

Fig. 3-3-12

(10) MAG PATH (MAXIMUM GAIN PATH)

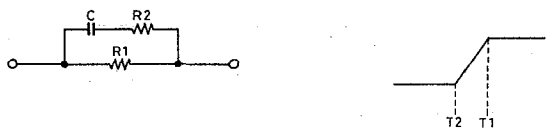
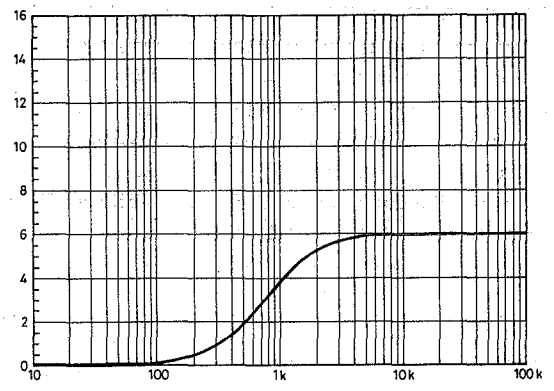


Fig. 3-3-13

$T_1 = CR_2 \approx 150\mu sec$ (1.06kHz)

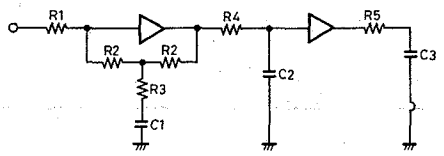
$T_2 = C(R_1 + R_2) \approx 300\mu sec$ (0.531kHz)



SNR MAG CCIR

Fig. 3-3-14

(11) WEIGHTING



$$\frac{2R_2}{R} \cdot \frac{1 + SC_1(R_3 + R_2/2)}{1 + SC_1R_2} \cdot \frac{1}{1 + SC_2R_4} \cdot \frac{SC_3}{1 + SC_3R_5}$$

$$= \frac{2R_2}{R_1} \cdot \frac{1 + ST_1}{1 + ST_2} \cdot \frac{1}{1 + ST_3} \cdot \frac{1}{1 + ST_4}$$

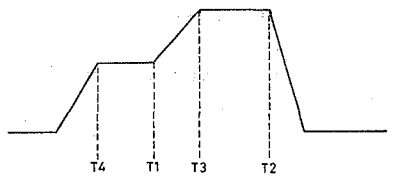


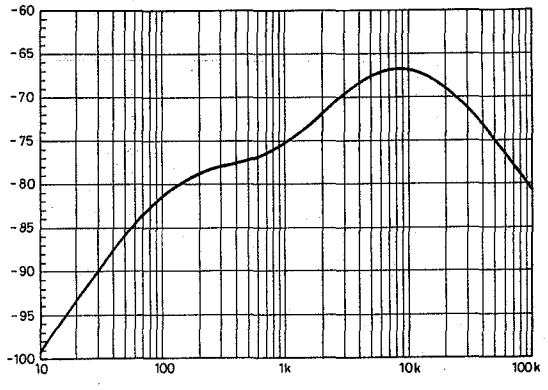
Fig. 3-3-15

$T_1 = C_1(R_3 + \frac{R_2}{2}) \approx 158\mu sec$ (1.00kHz)

$T_2 = C_1R_3 \approx 12.2\mu sec$ (13.0kHz)

$T_3 = C_2R_4 \approx 39.0\mu sec$ (4.08kHz)

$T_4 = C_3R_5 \approx 1.34msec$ (119Hz)



SNR WEIGHTING CCIR

Fig. 3-3-16

(2) FREQUENCY MODULATION BLOCK

A. REC mode operation description

The left (right) channel audio signal compressed by the SNR circuit is input to pin 48 (pin 3) of IC2208, which is the modulator input. The DC component of this signal is cut by C2239 (C2427) and the amount of modulation is adjusted by R2259 and R2355 (R2235 and R2354).

The 1.44-MHz (2.1-MHz) left (right) channel carrier frequency is determined by the current supplied to pin 48 (pin 3) from the regulator output pin 2.

The carrier frequency is adjusted at SVR2210 (SVR2211) by adjusting the amount of current supplied from pin 2.

In addition, so that the carrier frequency will not have thermal characteristics, metallized components are used for SVR2210 (SVR2211), R2261 (R2232), and R2260 (R2231). The audio signal input to demodulator 1 (demodulator 2) has its 1.44-MHz (2.1-MHz) carrier frequency modulated. At the standard 33% modulation for an output of −10 dBs, the FM deviation is 66.7 kHz.

In addition, the maximum deviation at 100% modulation, as determined by the PAL Beta hi-fi format, is ±200 kHz. This is limited by the SNR IC's final-stage amplitude limiter so that it never exceeds 100%. The frequency-modulated carrier is output from pin 42 (pin 7) of IC2208 via an AGC amplifier, and then shaped into a sine wave by a 1.44-MHz (2.1-MHz) low-pass filter.

This signal is then input to pin 39 (pin 9) of IC2208 to control the AGC amplifier via the detector and maintain the demodulator output at a constant level.

The detector level is determined by the voltage applied to pin 41, and this pin 41 is biased by a circuit inside the IC. Although the demodulator output can also be varied by externally applying a bias voltage, this is not done here. For the demodulator levels, which are maintained at a constant level by the AGC amplifiers, first the left and right channels are mixed by Q2201, and then the resulting signal is supplied to the recording amplifier on the AD-3 circuit board.

IC2208 (CX20104) was developed for use in both NTSC format and PAL format Beta hi-fi VCRs. When used in the NTSC format, pin 43 (pin 6) is the PLL terminal; in the PAL format this pin is not used. Also, pin 38 (pin 12) is the capacitor terminal for the AGC loop filter.

REC MODE

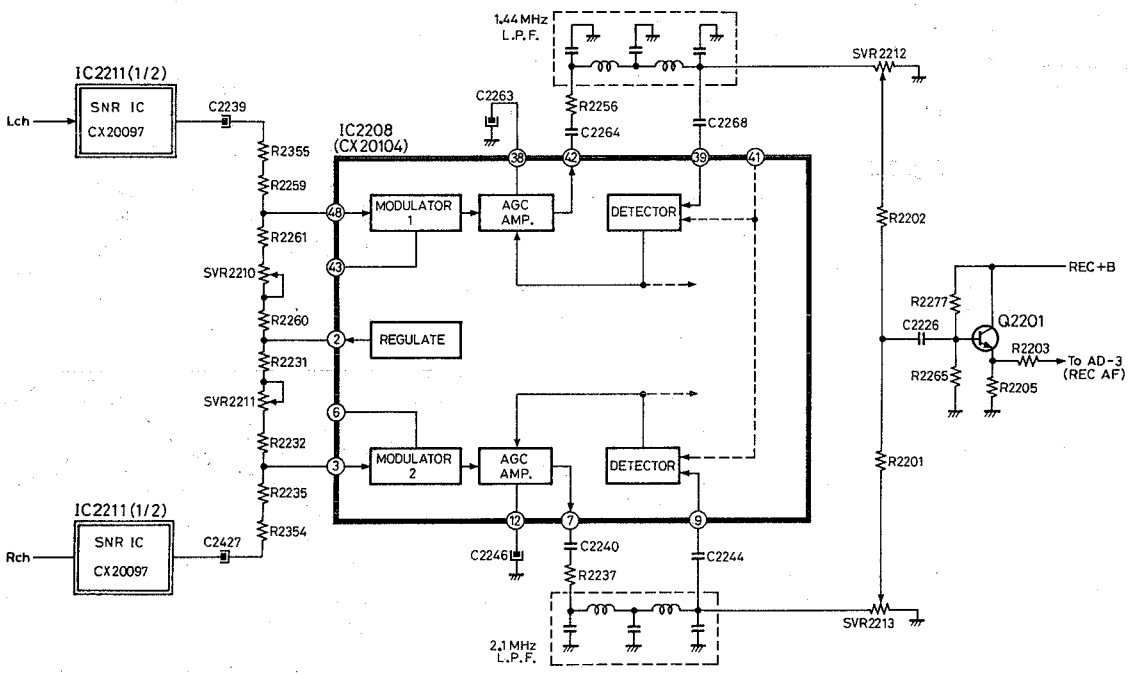


Fig. 3-3-17

B. PLAY mode operation description

The playback RF signal from the preamplifier on the AD-3 circuit board includes both the left (1.44 MHz) and right (2.1 MHz) channel carrier waves, so that F2206 and F2205 bandpass filters extract only the left (right) channel, which is then supplied to pins 26 and 28 (21 and 24). Because the level of this bandpass filter output is very small at only 10 mVp-p, it is first amplified approximately 100 times (40 dB) by the playback amplifier, and then supplied to the IC2209 (CX20105) and IC2210 (CX20105) demodulators.

Although the right channel undergoes the same signal processing as the left channel, if there is drop-out in the right channel, drop-out detection must be performed in order to compensate it.

The reason for performing drop-out detection for the right channel is because the right channel's carrier frequency of 2.1 MHz is higher than the left channel's 1.44 MHz, so the detection speed is slightly faster for the right channel.

For drop-out detection, the output of the left channel playback amplifier is shaped into a continuous RF envelope waveform by switch 1 and supplied to the VCA (voltage-controlled amplifier).
The switching of switch 1 is done by the audio FM switch pulse input to pin 19. The VCA absorbs variations such as in the tape, heads, or preamplifier circuit and supplies a stable RF level to the drop-out detector in the next stage.
The gain of the VCA is determined by using SVR2201 to adjust the voltage applied to pin 18. The terminal for measuring the output at this time is pin 13. Also, this terminal is normally in a saturated state.
The drop-out detector detects the drop-out and outputs a positive pulse corresponding to the drop-out period. This positive pulse is used for compensation of the pre-hold and muting performed by the demodulator.
When there is drop-out, it usually appears as RF envelope drop-out. In the deep-layer recording system, because the video signals are recorded in the shallow part of the tape's magnetic surface and the audio FM signals are recorded in the deep part, drop-out in the area where the video signals are recorded will cause the output level of the audio FM signal to increase. Consequently, drop-out compensation is performed for signal drop-out that reaches the audio FM recording layer. As shown in Fig. 3-3-18, the detection level of the drop-out detector is adjusted to approximately the limit at which the demodulator can demodulate.

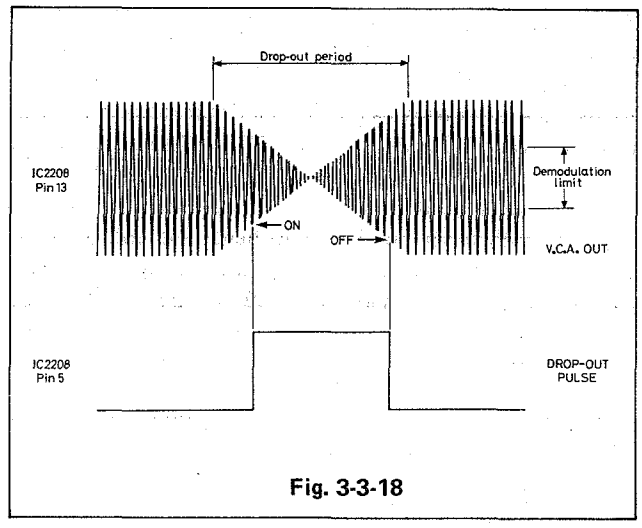


Fig. 3-3-18

Because the drop-out detection level is adjusted by changing the detector input, it is done by the VR2201 VCA gain control in the preceding stage. Also, the detector's threshold level is determined by the input voltage of pin 10 of IC2208.
In addition, a difference in the ON-OFF levels is provided in order to completely cover the drop-out period. The SSB converter circuit and 150-kHz oscillation circuit of IC2208 (CX20104) is only for NTSC format Beta hi-fi VCRs, and is not used in the PAL format.

PLAY MODE

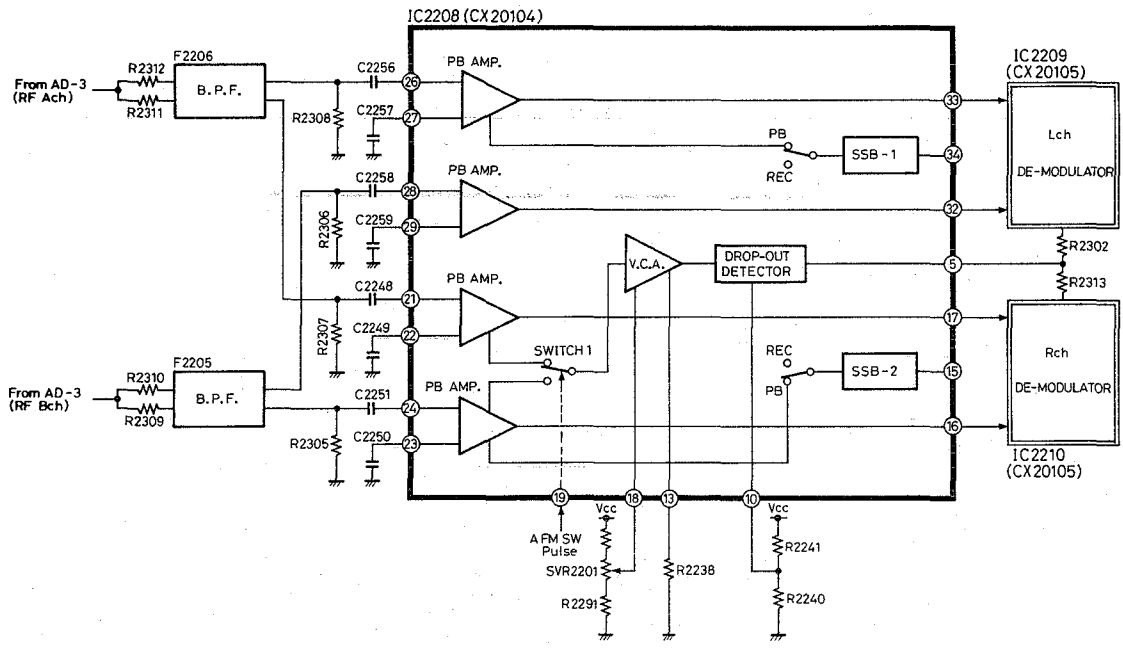


Fig. 3-3-19

(3) AUDIO FREQUENCY DEMODULATION BLOCK

A. Concerning the demodulators (two circuits per channel)

hi-fi VCRs are considerably different from conventional VCRs. The difference lies in the fact that while the fixed-position head in a conventional VCR produces continuous audio signals, a hi-fi VCR uses two rotating heads. These rotating heads (audio hi-fi heads) are mounted directly opposite each other on the video head disc and rotate at 25 Hz (1500 rpm). In addition, because the tape is helically wound around more than half (188° – 189°) of the video head drum, the audio FM signals become non-continuous every 50 Hz.

For the video signals, because the switching point is set within the vertical blanking period every 50 Hz, there is no problem with continuity. However, for the audio hi-fi signals, because the audio signals being input to the circuit are continuous, if the RF signal is used for switching, the phases of the FM signals would; not connect and a squeaking noise would be heard immediately after the switching of the audio output. For there reasons, the fact that the recording overlaps is utilized and two demodulators are provided for each audio channel.

With this system, by staggering the switching timing of the demodulators, we can obtain noise-free demodulation output. In addition, the demodulators used are pulse-count frequency demodulators.

Now we will explain about these pulse-count frequency demodulators.

The audio FM carrier frequencies are 1.44 MHz (left channel) and 2.1 MHz (right channel), which are lower than the carrier frequency of an FM tuner. In addition, the frequency deviation is set to a wide range of ±200 kHz.

Thus, if the PLL and quadrature modulation systems used in FM tuners were used in an audio hi-fi VCR, little could be expected in the way of linearity. Because the pulse-count demodulator does not make comparisons with a reference signal as other detectors do, it has good linearity even in wide band ranges. In addition, because the carrier frequency is low, it is easy to generate a good pulse, and there are also advantages concerning the S/N ratio. Thus this type of demodulator is ideal for use as an audio FM demodulator in hi-fi VCRs.

B. Demodulator operation description

The FM signals amplified by IC2208 (CX20104) are input to the IC2209 (left channel) and IC2210 (right channel) demodulators. Because IC2209 and IC2210 are the same IC, our explanation here will be only for IC2210 (right channel).

The audio FM signals are input to pins 42 (channel B) and 2 (channel A) of IC2210. They both pass through limiters where the AM components are removed, and then they are supplied to switches 1 and 2.

Meanwhile, delay pulses a, b, and c are generated by the TPG (timing pulse generator) from the audio FM switch pulse input to pin 39 of IC2210. These pulses control switches 1 and 2 in order to supply the FM signals to demodulators 1 and 2.

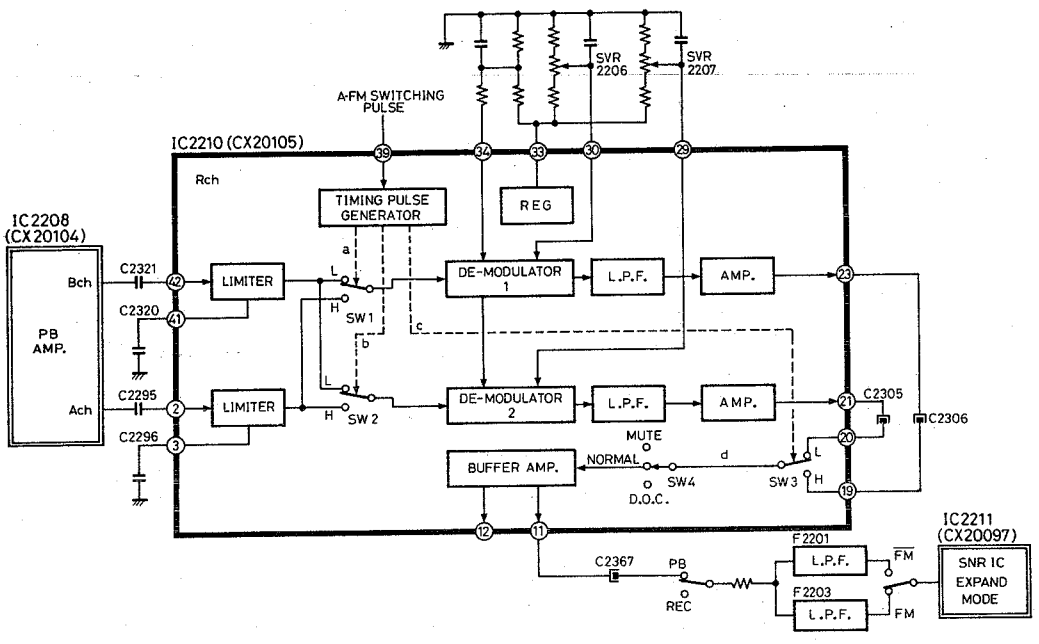


Fig. 3-3-20

As shown in Fig. 3-3-21, there is noise in the demodulator outputs for a period of several microseconds immediately after switching. Switch 3 connects the parts where there is no noise in order to obtain the noise-free outputs shown at d.
C2305 and C2306 are DC cut capacitors for eliminating the differences in the DC levels of the demodulator outputs.

PRINCIPLE OF TWO DEMODULATORS PER CHANNEL

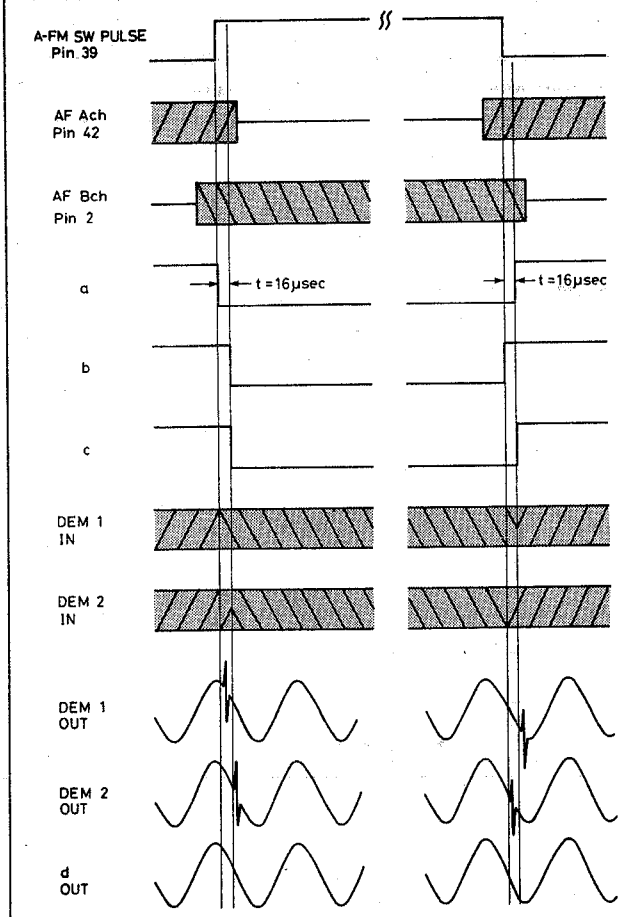


Fig. 3-3-21

Switch 4 is for drop-out compensation, and it is in the normal position during normal audio FM playback.
The output of switch 4 passes through a buffer amplifier and is output to pin 11. The signal from pin 11 IC2210 passes through a low-pass filter and is supplied to the SNR circuit. The cut-off frequency of this low-pass filter is 23 kHz, so only flat-response audio signals within 20 kHz pass through. In addition, any carrier leak or noise component above 20 kHz is removed because it would cause operation errors in the SNR circuit.
In IC2210, pins 30 and 29 determine the demodulation level of demodulators 1 and 2. A fixed bias is applied by the regulator output of pin 33 and SVR2206 and SVR2207 bias pins 30 and 29, which are the demodulation level terminals, in order to equalize the demodulator levels, thus achieving a demodulation balance between demodulators 1 and 2.

C. Drop-out compensation circuit and audio FM discrimination circuit

The most common drop-out length (period) is 1 to 2 H, and it usually occurs at random. Also, with the playback sound of standard music recording, drop-out of up to approximately 150 to 300 μsec. in length is virtually undetectable by the human ear even if only pre-hold compensation is done.

However, for longer drop-out of more than 300 μsec. the connection with the music source sound becomes weaker and the audio output is muted. In addition, if severe scratches on the tape cause the mute compensation to occur frequently, and also when a tape which has not been audio FM recorded in played back, there is no audio FM carrier, so the audio FM is muted.

When this happens, because the absence of the sound from the line output continues, the audio FM discrimination circuit identifies this condition and outputs a signal to the output switching circuit directing it to automatically switch to the normal audio mode. This audio FM discrimination circuit removes the noise (80 to 150 kHz) not within the audio band range from the audio FM demodulator output and makes use of the fact that the noise level when there is no audio FM signal is approximately 20 dB higher than that when there is an audio FM signal.

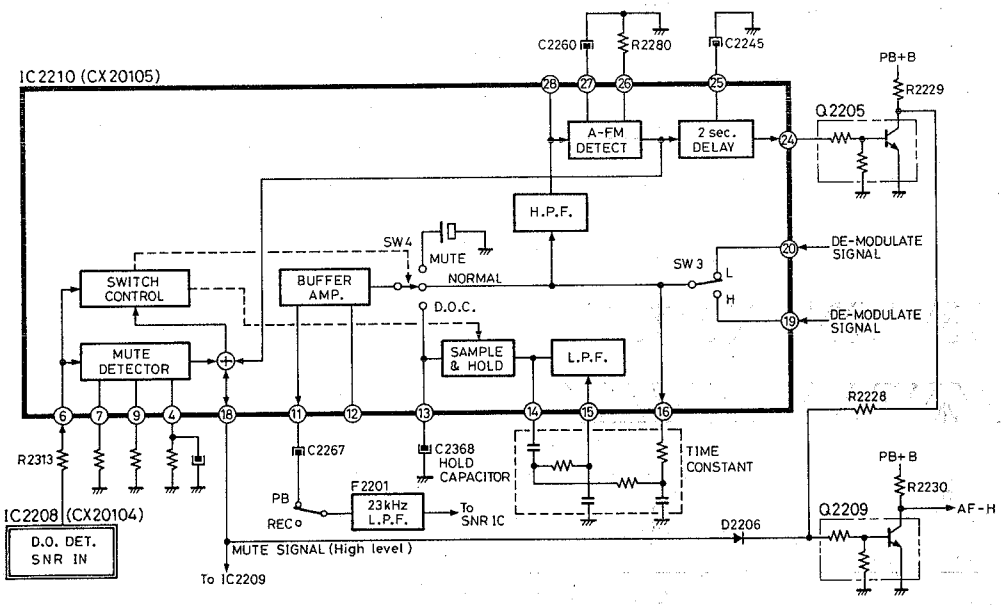


Fig. 3-3-22

The output from pin 5, which is IC2208's drop-out detection terminal, is HIGH level when there is drop-out. This HIGH level pulse signal is input to pin 6 of IC2210 (CX20105), and then supplied to the switch control and the mute detector. For drop-out too short to cause the mute detector to operate, the switch control switches switch 4 to the DOC (drop-out compensation) side in real time and sends the pre-hold instruction to the sample-and-hold circuit.

At this time, the demodulator output signal (audio signal) switched by switch 3 is output to pin 16, passed though the time constant of a low-pass filter, and input to pin 15. The audio signal input to pin 15 passes through a low-pass filter and is supplied to the sample-and-hold circuit. However, because the demodulator output generates pulse noise when there is drop-out, if the signal is sampled and held like this and there has been an error by the drop-out detector, the hold could be applied to the pulse noise, thus having the opposite effect of that desired. For this reason, the low-pass filter first integrates the signal to make the pulse noise smaller, and then the signal is supplied to the sample-and-hold circuit. C2368 (pin 13) is a capacitor used for this hold function.

Next, if the length of the drop-out reaches 300 μsec. the mute detector performs the mute processing operation. As shown in Fig. 3-3-23, the mute detector outputs the mute pulse when the drop-out pulse reaches 300 μsec. and it has a recovery time of approximately 100 msec. after the drop-out pulse is finished. This is to prevent the repeated switching ON-OFF of the muting operation from becoming annoying to hear.

In addition the mute detector output is sent to the switch control via pin 18's OR circuit (+) and causes switch 4 to switch to the mute side. Also, output to pin 11 is prohibited.

A mute detector is included both IC2209 for the left channel and in IC2210 for the right channel; however, in this model only the one in IC2210 (right channel) is used. Thus pin 18 of IC2210 is used for both input and output of the mute pulse, and the left channel mute pulse is supplied from the right channel so that the left channel functions in the same way as the right channel.

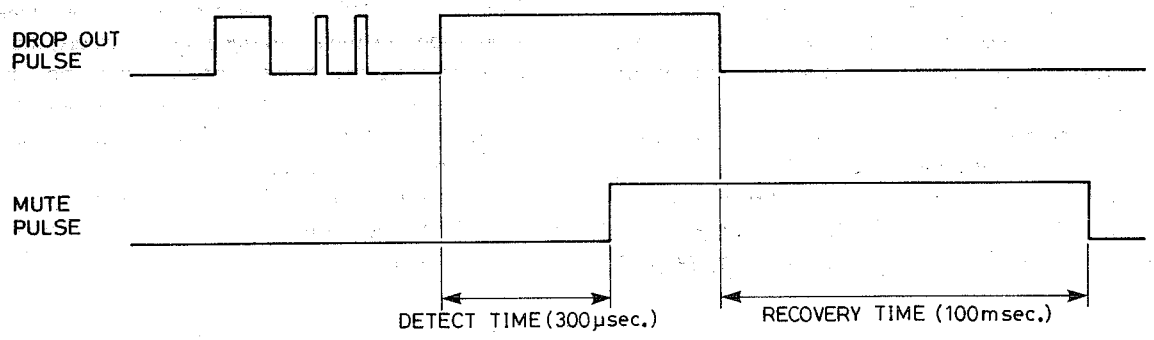


Fig. 3-3-23

D. Audio FM detector operation description

In the audio FM detector the demodulated audio signal (including when the playback audio signal has not been audio FM recorded) is limited to 150 kHz by a low-pass filter and then passed through switch 3. Next the signal is passed through a 80-kHz high-pass filter so that it is limited to a band of 80 to 150 kHz. The signal is detected by a full-wave rectifier, and then a comparator determines whether or not the A-FM signal has been recorded. The comparator output is divided into two signals: one causes the mute circuit to operate, and the other passes through a delay circuit where it is set to an appropriate detection speed in order to discriminate the A-FM signal.

The audio FM discrimination signal output from pin 24 of IC2210 is inverted by Q2205, and then inverted again by Q2209 so that the AF-H signal is supplied to the output switching circuit.

Pin 28 of IC2210 is the terminal which traps the 150-kHz carrier leak in the SSB converter circuit for the NTSC format. Pin 27 is a capacitor for smoothing the rectification output, and pin 26 is for setting the threshold level. Also, pin 25 is the capacitor terminal for the delay of audio FM discrimination.

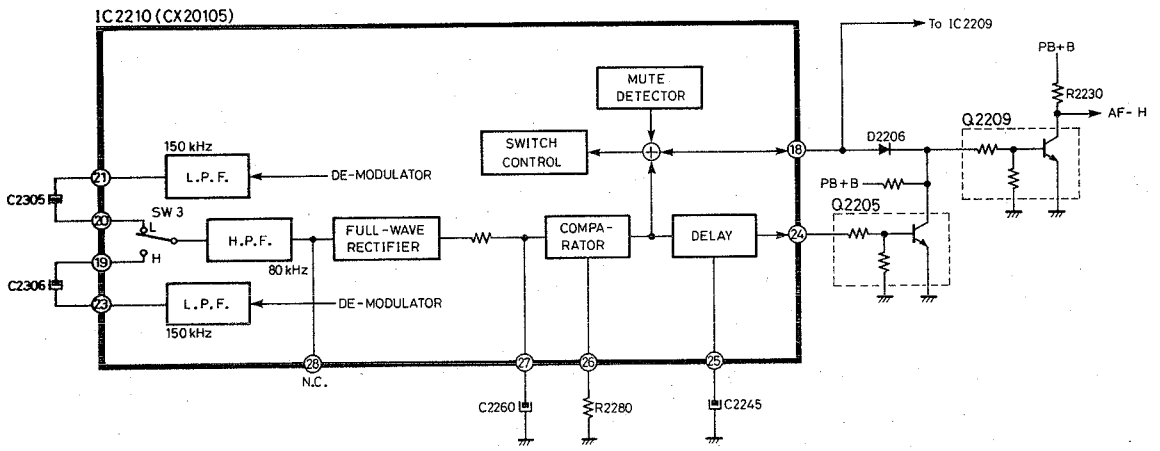


Fig. 3-3-24

4. NORMAL AUDIO AND AUDIO SIGNAL SELECT CIRCUITS

1. OUTLINE

The ICs which make up the normal audio circuit and the audio signal select circuit are shown in Table 4-1-1. Although the IC composition of the VTC-M40 series and the VTC-M50 series are different, the method of signal processing in the normal audio circuit is almost exactly the same as in previous models. Also, the main ICs (LA7042 and BA5115L) used in the normal audio circuit operate on basically the same fundamental principles.

The biggest difference between LA7042 and BA5115L is that while LA7042 incorporates the previous audio circuit and its peripheral circuits (mute control, REC/PB switch, etc.) into the IC itself, BA5115L can be viewed as a simplified version of the previous IC's internal composition. This difference in turn results in differences in the number of IC terminals and in the type of package (DIP, or dual-in-line package, versus SIP, or single in-line package).

Thus, because the basic operation of the normal audio circuit can be considered the same as in previous models, the explanation of that operation will not be repeated here.

In addition, for the audio signal select circuit, the ICs which make up the circuit handle virtually all of the signal switching functions for the audio FM and normal audio signals and for the external switches (audio output, stereo/mono, auto/manual level meter, etc.).

Thus here we will omit the explanation of operation related to the various external switches and discuss only the operation specifications of the ICs themselves.

	IC Name	M40 Location No.	M50 Location No.	Function
NORMAL AUDIO	LA7042	IC2010		NORMAL AUDIO SIGNAL PROCESS
	BA5115L		IC2206	NORMAL AUDIO SIGNAL PROCESS
AUDIO SELECT	TK15021	IC2001	IC2202, IC2203	ANALOG SWITCH
	TK15050		IC2201	ANALOG SWITCH
	LA3220	IC2002		ALC AMP.
	TC4066BP	IC2003, IC2005		QUAD BILATERAL SWITCH
	HD14052BP	IC2004		DUAL 4-ch ANALOG MULTIPLEXER/ DEMULTIPLEXER
	NJM4558S		IC2204	OP. AMP.
	TK15090Z		IC2205	ANALOG SWITCH
	LA3210		IC2207	ALC AMP.

Table 4-1-1

OVERALL WIRING DIAGRAM (NORMAL AUDIO CIRCUIT)

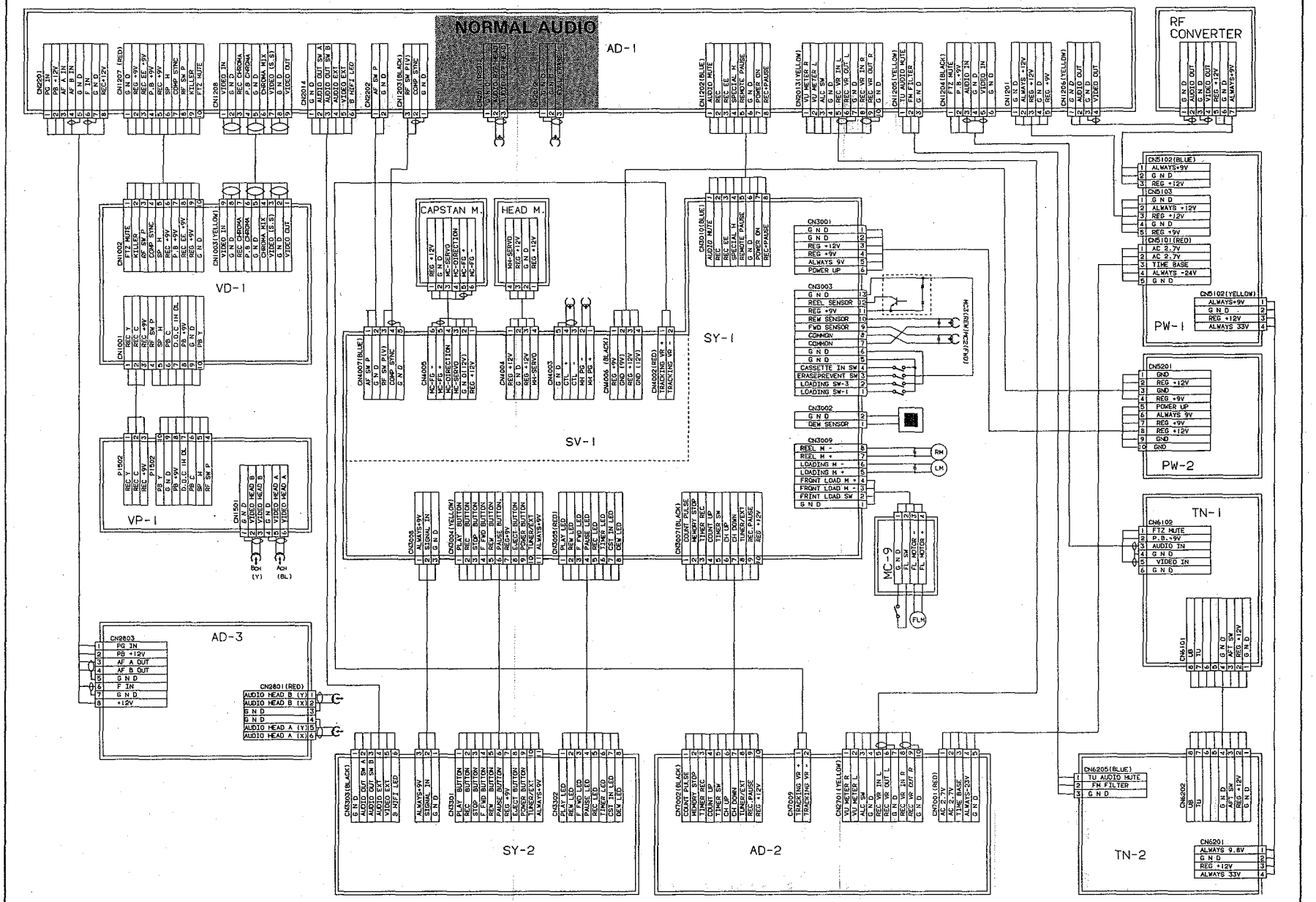


Fig. 4-1-1

OVERALL WIRING DIAGRAM (AUDIO SIGNAL SELECT CIRCUIT)

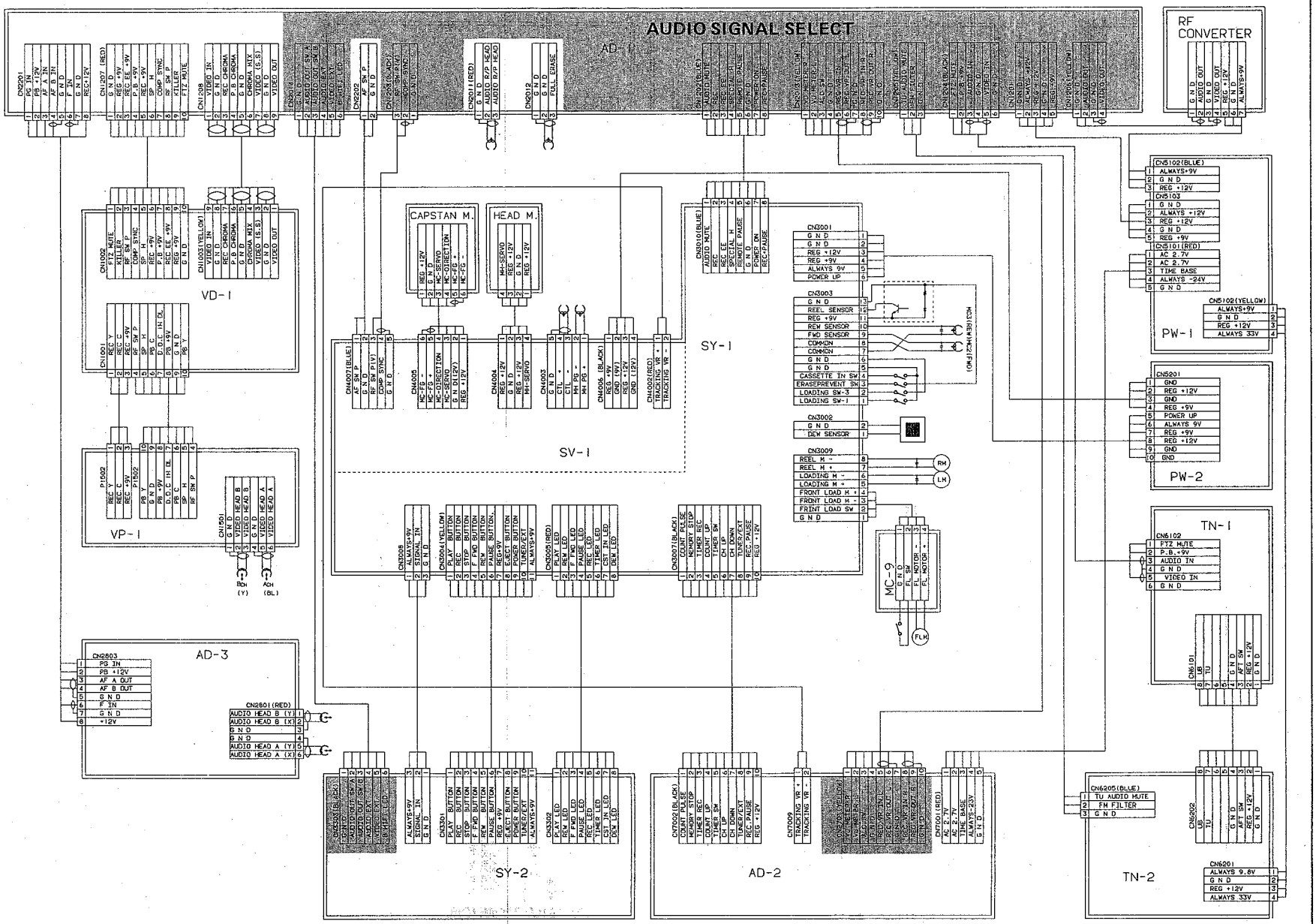


Fig. 4-1-2

2. DESCRIPTION OF THE ICS IN THE NORMAL AUDIO AND AUDIO SIGNAL SELECT CIRCUITS

We will now explain the specifications of the main ICs which make up the normal audio circuit and the audio select circuit.

(1) LA7042

LA7042 has been used in previous VCRs for recording and playback of the audio signal, and it includes the following functions.

- Microphone, equalizer, recording, line, and various other amplifiers.
- ALC circuit
- Recording/playback control circuit
- Mute control circuit

In addition, the IC also has the following features:

- Includes the recording and playback amplifiers needed in the VCR's audio signal system.
- Muting can be applied easily by applying a voltage to the control terminal.
- ALC is added to microphone and line amplifier inputs in a two-stage composition, thus achieving a good S/N ratio during recording.
- Shock noise from switching of REC/PLAY modes is extremely small.
- Includes a head switch for grounding of the tape head's playback output terminal during the REC mode. (Can be used together with the switch circuit for the recording head terminal with a discrete transistor.)

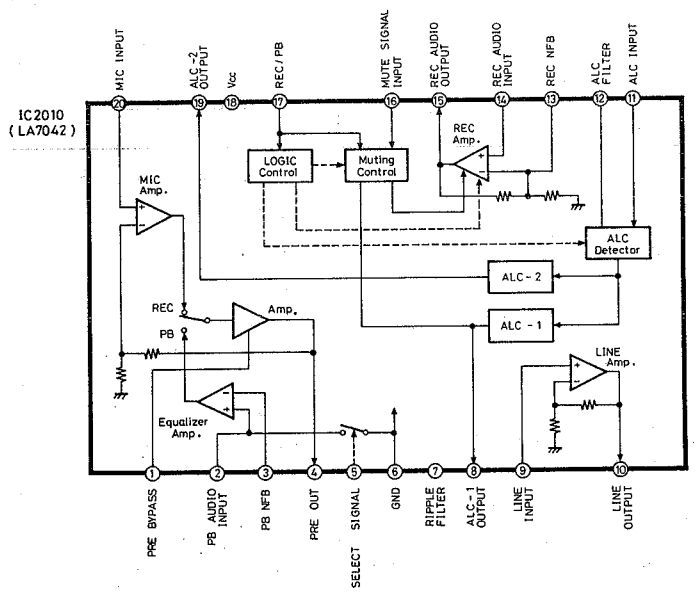


Fig. 4-2-1

In addition to the aforementioned features, LA7042 is also capable of control of the power supply voltage level. Figure. 4-2-2 shows the relationship between pin 17 (REC/PB) and the REC and PB modes when the power supply voltage (V_{CC}) at LA7042 is 9 V. Also, Fig. 4-2-3 shows the relationship between $V_{R/P}$ and the REC and PB modes according to the Vcc level.

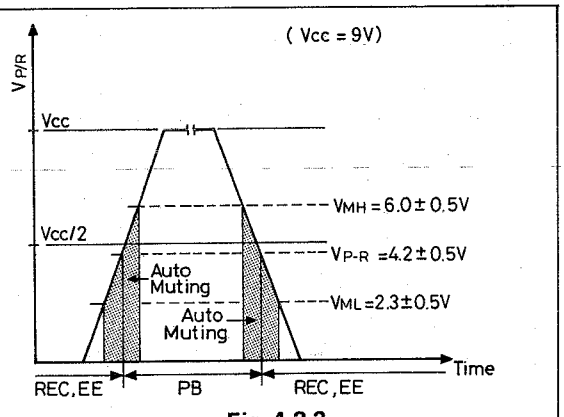


Fig. 4-2-2

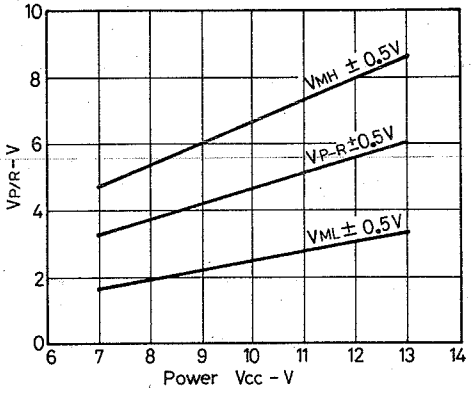


Fig. 4-2-3

In addition, LA7042's mute control function is illustrated by the V_{CC} —V_M characteristics in Fig. 4-2-4. This figure shows that the audio mute condition will be enforced when a voltage of V_M or above is applied to pin 16, and it also shows the relationship between the V_{CC} level and the V_M level.

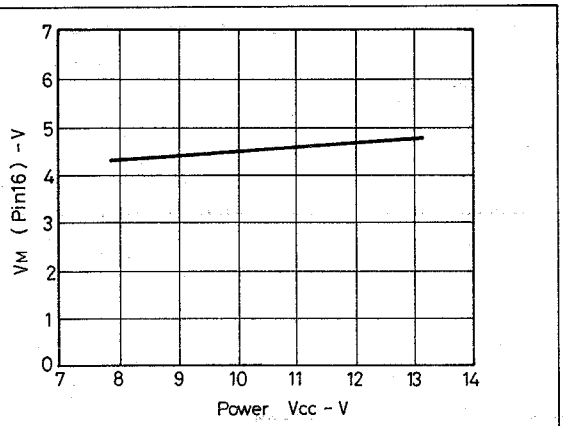


Fig. 4-2-4

(2) BA5115L

BA5115L is used as a switchless recording/playback amplifier for the VCR's audio signal. The IC incorporates the following functions.

- Recording, playback, line, buffer, and various other amplifiers
- ALC circuit
- REC/PB/EE control circuit

In addition, the IC also has the following features.

- Control terminals for REC/EE, PB/EE, and line muting, thus making interface with the control system easier. (The REC/EE terminal is also capable of audio dubbing control.)
- The use of a high-level ALC control provides an excellent S/N ratio and distortion ratio.
- ALC level can be set by externally connected resistor.
- Extremely low occurrence of popping noise at initial supply of power and at disconnection of power supply.
- Extremely low occurrence of shock noise from switching of the control system.

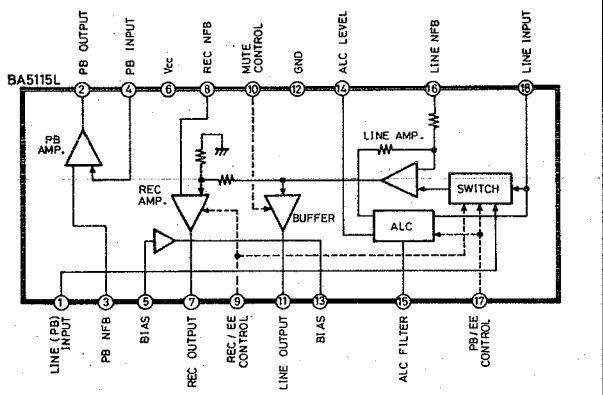


Fig. 4-2-5

The control functions of BA5115L are shown in Table 4-2-1.

a. Control terminal functions

Control terminal	Control functions
REC/EE CTRL	●REC: HIGH; EE: LOW ●ON/OFF switching of recording amplifier. ●This terminal takes precedence over PB/EE control terminal, so it can be used for the AUD DUB (Audio dubbing) mode.
PB/EE CTRL	●BB: HIGH, EE: LOW ●Control of ALC ON/OFF and selection of REC/PB input at line amplifier input.
MUTE CTRL	●Mute: HIGH ●ON/OFF switching of line output. ●Capable of control independent of the ALC loop recording amplifier.

b. Control modes

Mode	Control system			Function				
	REC/EE	PB/EE	MUTE	EQ AMP.	LINE SW	LINE OUT	REC OUT	ALC
EE	L	L	L	(ON)	REC	ON	OFF	ON
EE MUTE	L	L	H	(ON)	REC	OFF	OFF	ON
REC	H	L	L	(ON)	REC	ON	ON	ON
REC MUTE	H	L	H	(ON)	REC	OFF	ON	ON
PB	L	H	L	ON	PB	ON	OFF	OFF
PB MUTE	L	H	H	ON	PB	OFF	OFF	OFF
AFR	H	H	L	(ON)	REC	ON	ON	ON
AFR MUTE	H	H	H	(ON)	REC	OFF	ON	ON

Table 4-2-1

(3) TK15021

TK15021 is used as an analogue switch in the audio circuit system. The IC contains two two-position switch circuits with independent control terminals (SELECT INPUT). When the SELECT INPUT terminals are in the normal state (HIGH level), channel A is output.

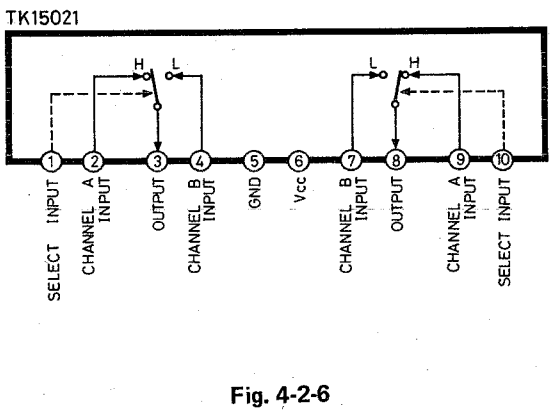


Fig. 4-2-6

(4) TK15050

TK15050 is used as an analogue switch in the audio circuit system. The IC contains two two-position switch circuits which are interlock-operated by the toggle system control input and two three-position switch circuits which are interlock-operated by the control 3 inputs.

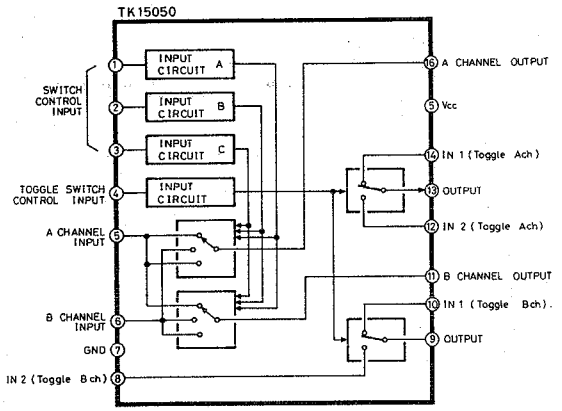


Fig. 4-2-7

(5) LA3220

LA3220 is a dual preamplifier which contains the ALC function for a stereo radio/cassette player. The IC has the following features.

- Capable of variable monitoring.
- Capable of direct meter drive and ALC.
- Good channel separation, voltage-reduction characteristics, and balance of the ALC characteristics.

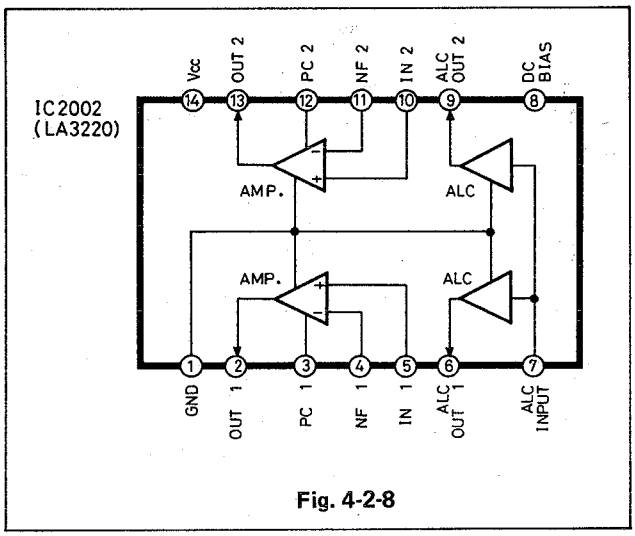


Fig. 4-2-8

(6) TC4066BP

TC4066BP contains four independent bilateral switch circuits. Fig. 4-2-9 shows the IC's logic diagram and truth table. In this IC, when the C_{IN} control input is HIGH level, there is low impedance between the switch inputs/outputs; when the C_{IN} control input is LOW level, the impedance is high. The ON and OFF resistances (R_{ON} and R_{OFF}) for the impedance between the switch inputs/outputs are as follows:

ON resistance (R_{ON})

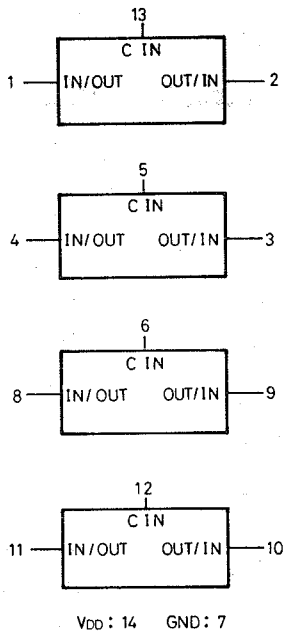
300Ω (typical): $V_{DD} - V_{SS} = 5$ V

110Ω (typical): $V_{DD} - V_{SS} = 10$ V

70Ω (typical): $V_{DD} - V_{SS} = 15$ V

OFF resistance (R_{OFF})

R_{OFF} (typical) $> 10^9 \Omega$



TRUTH TABLE

C IN	Impedance Between IN / OUT - OUT / IN
H	$0.5 \sim 5 \times 10^2\ \Omega$
L	$10^9\ \Omega$

Fig. 4-2-9

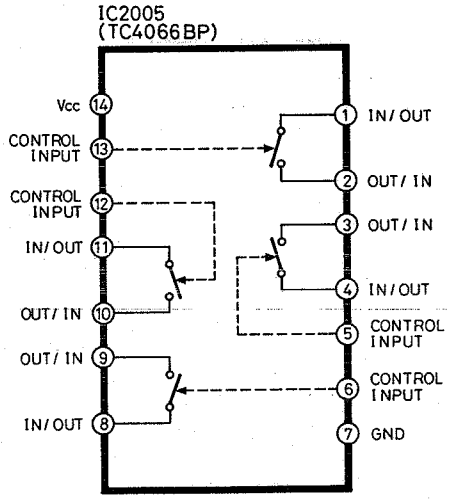
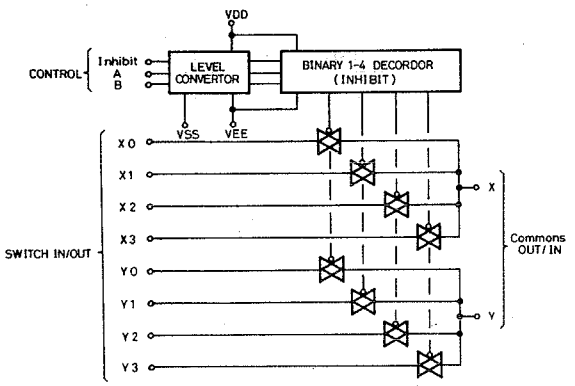


Fig. 4-2-10

(7) HD14052BP

HD14052BP contains two four-channel analogue type multiplexer/demultiplexer circuits, and switches signals by using three control inputs, including an inhibit input. The IC's logic diagram and truth table are shown in Fig. 4-2-11.

LOGIC DIAGRAM



TRUTH TABLE

CONTROL INPUT			ON SWITCH	
Inhibit	B	A		
L	L	L	Y0	X0
L	L	H	Y1	X1
L	H	L	Y2	X2
L	H	H	Y3	X3
H	※	※	※	※

H : High level
L : Low level
※ : Don't care

Fig. 4-2-11

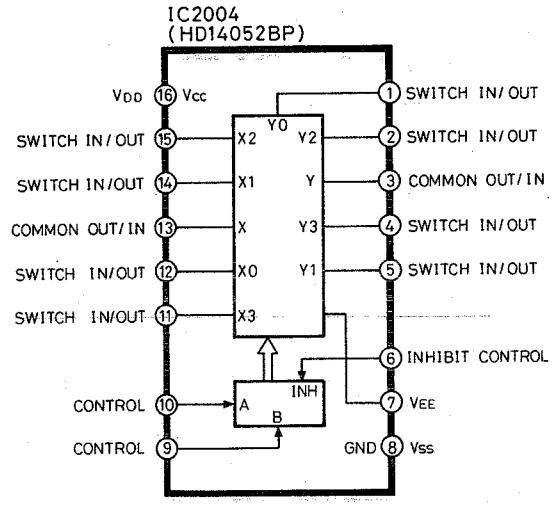


Fig. 4-2-12

(8) TK15090Z

TK15090Z is an analogue switch circuit for the hi-fi VCR audio output. The IC contains an amplifier circuit, a mixer circuit, and a logic circuit. Refer to the circuit diagram in Fig. 4-2-13 for the control operation of TK15090Z. Also, the switch control input level voltages when TK15090Z is V_{CC} (+8 V) are shown in Table 4-2-2.

Reference Vcc = +8V

Pin No.	Level	Level voltage
10	Low	−0.3 ~ 0.8
	Center	1.6 ~ 3.4
	High	4.2 ~ 5.0
11	Low	−0.3 ~ 1.4
	Center	2.0 ~ 3.0
	High	4.2 ~ 5.0
7, 12, 13, 14	Low	−0.3 ~ 0.35
	High	1.8 ~ 5.0

Table 4-2-2

(a) The control voltage input to pin 10 (switch control) in Fig. 4-2-13 causes the signals shown in Table 4-2-3 to be output from pins 16 and 18.

Pin 10 level / Pin No.	Low (0 ~ 0.8V)	Center (1.6 ~ 3.4V)	High (4.2 ~ 5.0V)
16	Rch	Rch	Lch
18	Rch	Lch	Lch

Table 4-2-3

(b) The conditions of pins 12 and 14 (switch control) in Fig. 4-2-13 cause the signals shown in Table 4-2-4 to be output from pins 15 and 17 (VU meter output).

SWITCH	Pin 12 (SW2)	Open	Low	—
CONTROL	Pin 14 (SW4)	Open	Open	High
VU meter	Pin 15	Same signal as pin 16	Signal B	Same signal as pin 16
OUT signal	Pin 17	Same signal as pin 18	Signal A	Same signal as pin 18

Table 4-2-4

(c) Table 4-2-5 shows the output signals of pins 16, 18, and 19 Fig. 4-2-13 when TK15090Z is controlled by SW2 through SW6. Also, with the exception of MUTE and MONO, the output signals of pins 16 and 18 in Table 4-2-5 are determined by the control voltage of pin 10, as mentioned in section (a) above.

SWITCH CONTROL	Pin 14 (SW4)	Open	Open	High	High	High	High	High	High	High	High
	Pin 13 (SW3)	—	—	Open	Open	Low	Low	Low	Open	Open	Open
	Pin 12 (SW2)	Low	Open	Low	Low	—	—	—	Open	Low	Open
	Pin 11 (SW5) (SW6)	—	—	High	Center	Center	High	Low	Low	Low	High Center
CONTROL OUTPUT	Pin 16, Pin 18	Table 4-2-3	MONO	Table 4-2-3	MUTE	MONO	Table 4-2-3 + MONO	MONO	MONO	MONO	MONO
INDICATOR ON/OFF	Pin 19	ON	OFF	ON	ON	ON	ON	OFF	OFF	OFF	ON

Table 4-2-5

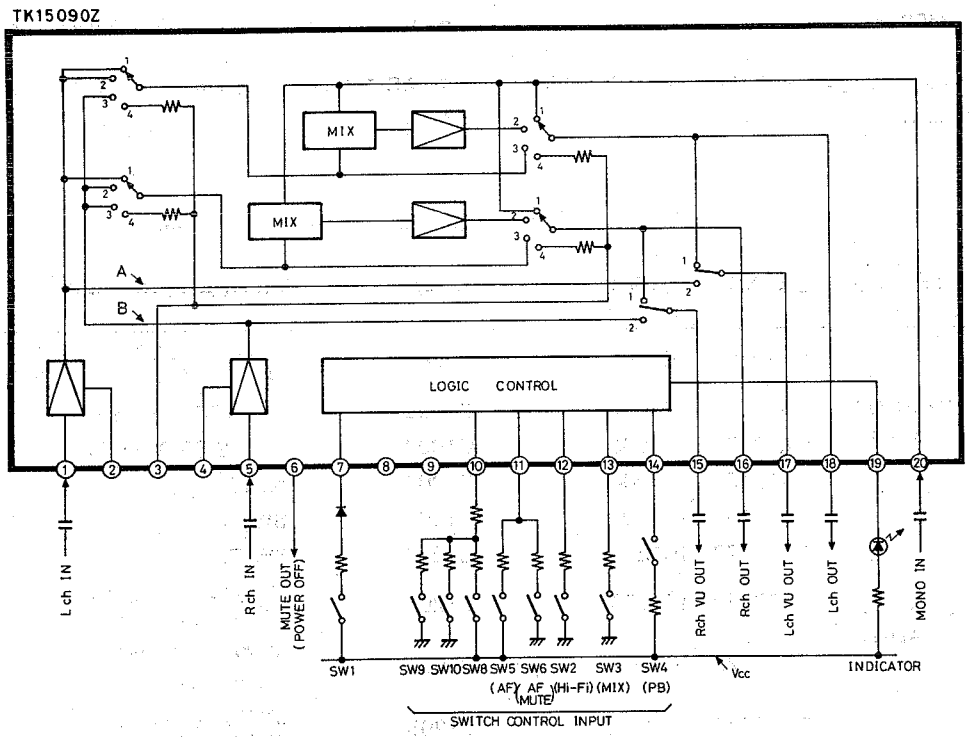


Fig. 4-2-13

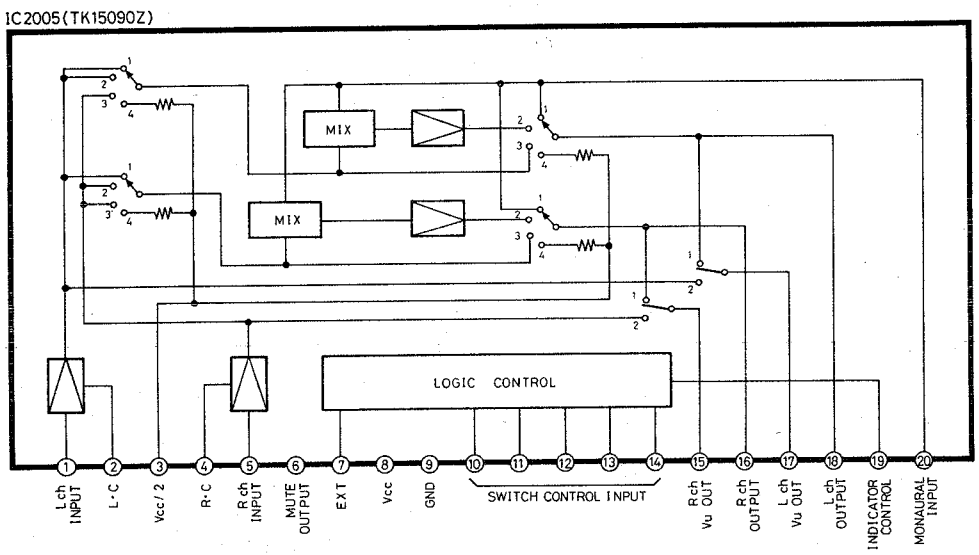


Fig. 4-2-14

5. VIDEO CIRCUIT

1. OUTLINE

The ICs used in the video circuit are, as shown in Table 5-1-1, the same as those used in the VTC-M40 series and the VTC-M50 series, and circuit operation is also the same In addition, the video circuit has, for the most part, the same circuit operation principles as the former PAL Beta system, in order that there will no effect caused by the PAL Beta hi-fi system for audio FM.

Consequently, a detailed description of the video circuit operation is omitted from this manual, as is the explanation related to the VTC-M50 series.

This manual will describe the newly employed ICs, with the VTC-M40 as the basic model, and the main ICs which were used formerly as well.

The new ICs are mainly for the chroma signal processing circuit. Some of the peripheral circuits which were of discrete configuration is former circuits are now included within the ICs, making these ICs multi-functional.

The video circuit of model VTC-M40 is composed of a portion of the AD-1 board, the VD-1 board, and the VP-1 board (shown in Fig. 5-1-1).

The main functions of each of the boards of the video circuit are as described below.

- VP-1 board (recording/playback amplifier of video RF signals)
- VD-1 board (recording/playback processor of luminance signals)
- AD-1 board (recording/playback processor of chroma signals)

	IC Name	M40 Location No.	M50 Location No.	Function
VIDEO Pre-AMP. (VP-1)	LA7007S	IC1501	IC1501	PB HEAD PRE-AMP.
Y•SIGNAL (VD-1)	LA7210	IC1001	IC1001	SYNC. SIGNAL DETECTOR
	LA7020	IC1002	IC1002	Y • REC PROCESS
	LA7025	IC1003	IC1003	Y • PB PROCESS
	LA7013	IC1004	IC1004	NOISE CANCEL
C•SIGNAL (AD-1)	LA7016	IC1201	IC1201	SWITCHING
	TA7347P	IC1202	IC1202	SWITCHING
	TA7788AP	IC1203	IC1203	REC/PB CHROMA PROCESS
	TA7722P	IC1204	IC1204	SP PB CHROMA PROCESS
	TA7357AP	IC1205	IC1205	SYNC. SIGNAL SEPARATOR
	BA7007		IC1401	PAL/SECAM SIGNAL DISCRIMINATION

Table 5-1-1

OVERALL WIRING DIAGRAM (VIDEO CIRCUIT)

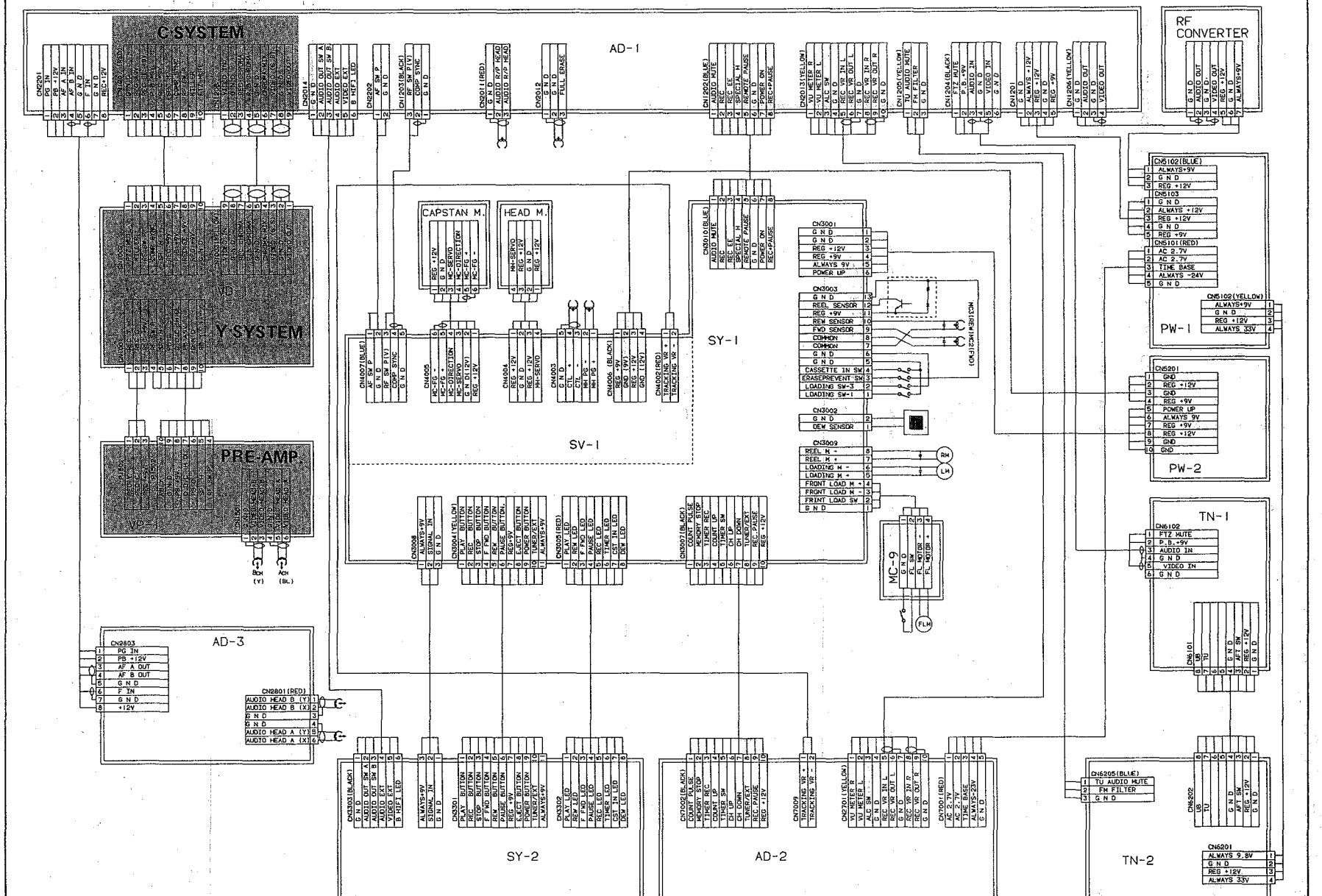


Fig. 5-1-1

2. DESCRIPTION OF VIDEO CIRCUIT ICS

The following is a description of the main ICs used in the video circuit.

(1) LA7007S

The LA7007S is an IC which has been used in the past as well for the playback pre-amplifier of VCRs. It includes the following functions.

- Playback head amplifier
- RF switcher
- Monochrome/color switch
- Limiter
- Drop-out compensator
- Burst compressor

The features of this IC are as follows.

- The head amplifier has a 2-channel configuration.
- The operation level of drop-out compensation can be set to the desired level.
- A – 6 dB burst compensator is built in.

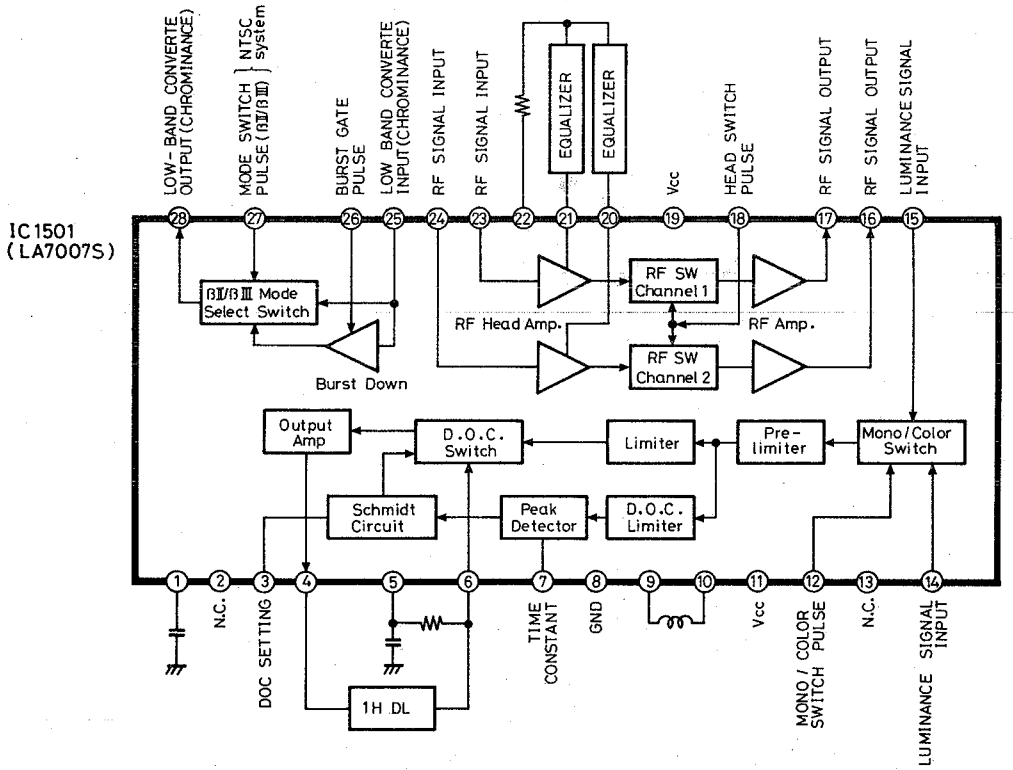


Fig. 5-2-2

(2) LA7210

The LA7210 is used in VCRs for sync-signal detection of video signals.

The LA7210 includes the following functions.

- Sync separator
- Comparator
- 32 fH VCO generator
- 1/32 down counter
- AFC

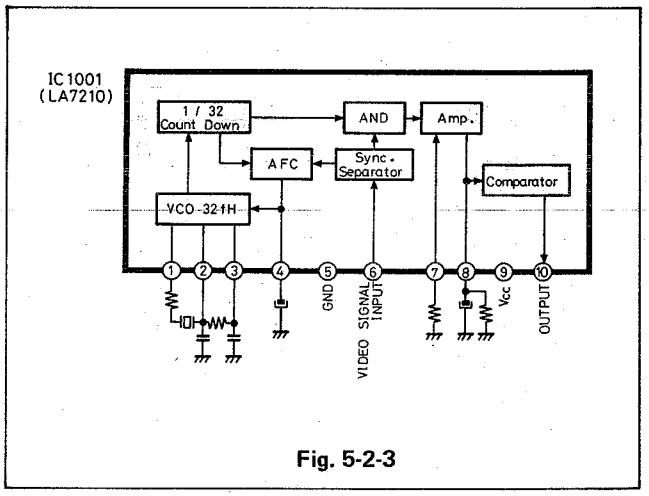


Fig. 5-2-3

(3) LA7020

The LA7020 is used, for PAL-system VCRs, in the luminance signal recording circuit.

The LA7020 includes the following functions.

- Sheathed AGC amplifier
- E-E amplifier
- Buffer amplifier
- Clamp circuit
- Sync separation
- Pulse delay
- Pre-emphasis
- White clipping
- Dark (black) clipping
- AGC (sync detection, peak detection)
- Frequency modulation circuit
- Limiter

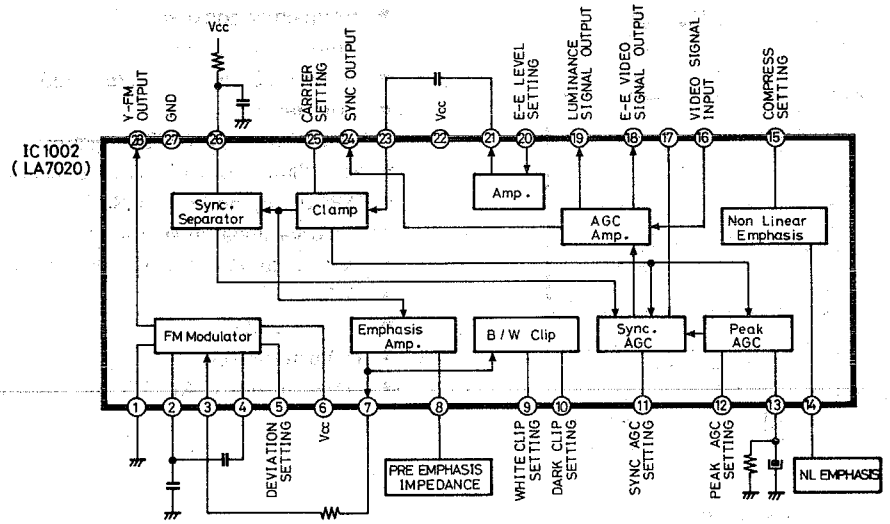


Fig. 5-2-4

(4) LA7025

The LA7025 is used, for PAL-system VCRs, in the luminance signal playback circuit.

The LA7025 includes the following functions.

- Demodulation circuit
- De-emphasis amplifier
- Non-linear emphasis
- Aperture-correction amplifier
- Sync-separation circuit
- White clipping/VD insertion
- Video signal clamping
- Luminance/chrominance signal mixing circuit
- Chroma signal amplifier
- Noise canceller
- Amplifier

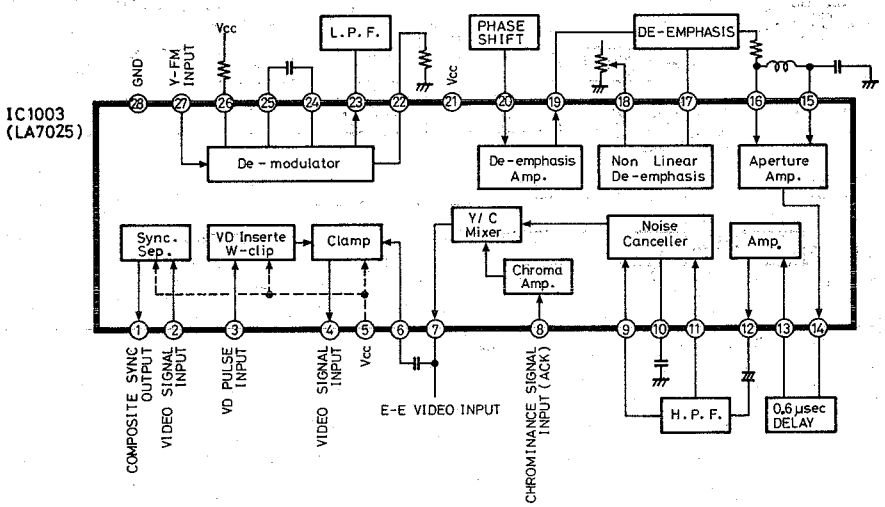


Fig. 5-2-5

(5) LA7013

The LA7013 is used for the luminance signal noise-canceller in VCRs.

The LA7013 includes the following functions.

- Frequency demodulator
- Adder
- Limiter

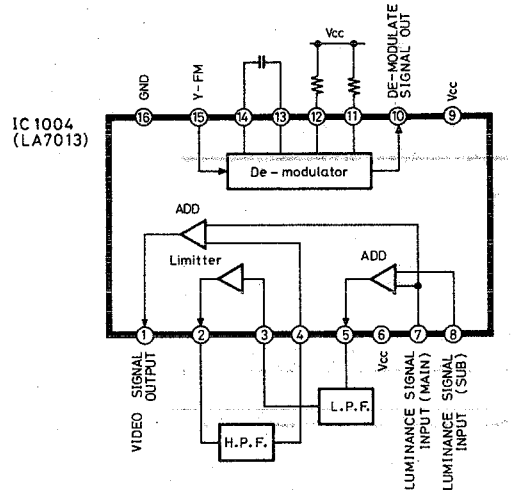


Fig. 5-2-6

(6) TA7788AP

The TA7788AP is an IC developed for chroma signal processing in NTSC/PAL/SECAM VCRs.

The TA7788AP includes the following functions.

- ACC input switching
- ACC amplifier
- ACC peak detection
- Pilot burst adder (PAL system)
- Burst emphasis (NTSC system)
- Frequency conversion (1)
- Frequency conversion (2)
- N•fH VCO (N = 350, 351, 353)
- Counter
- Phase inversion (NTSC system)
- AFC detection (NTSC/PAL systems)
- APC-ID detection (NTSC/PAL system)
- Chroma signal inversion (NTSC/PAL systems)
- ACK detection (NTSC/PAL systems)
- ACK
- 3 x fsc VCO
- HD generation
- N•fH output (PAL/SECAM systems) N = 351, 353

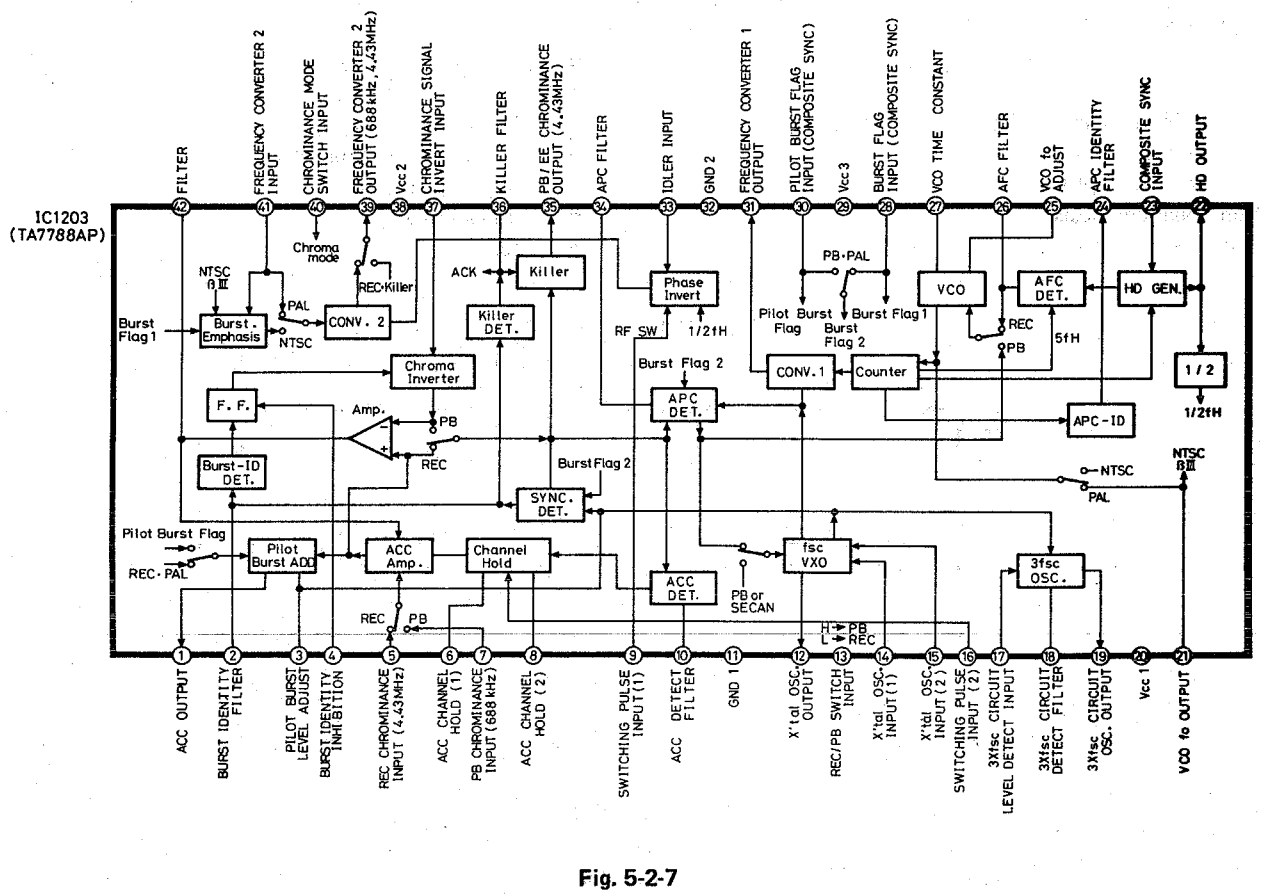


Fig. 5-2-7

(7) TA7722P

The TA7722P is an IC developed for special playback processing in PAL system VCRs.

The TA7722P includes the following functions.

- Jump detection
- Delay drive
- Variable gain amplifier (for 1H delay signals)
- Variable gain amplifier (for continuous wave signals)
- CW signal phase inversion
- CW signal phase composition
- HD pulse shaping
- fH VCO
- AFC detection
- BG generation
- Flip-flop
- Switching (direct signal/delay signal switching, exchange of burst signal, pilot burst cleaning, blanking period cleaning)

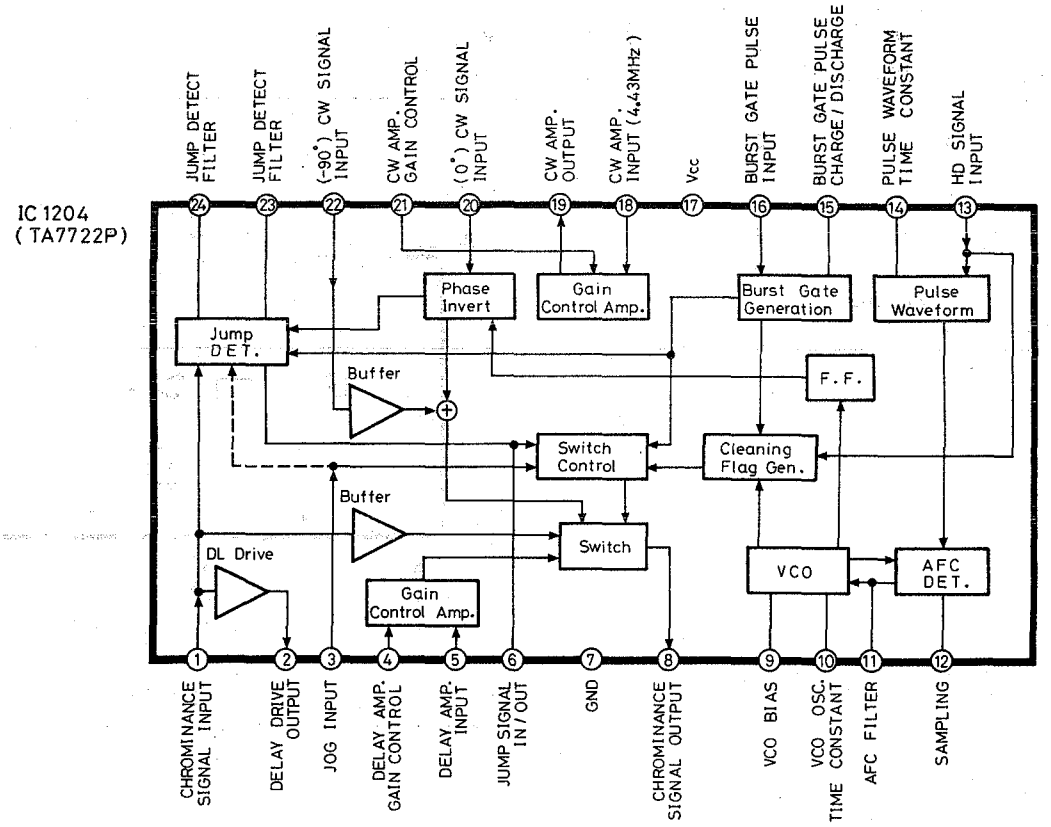


Fig. 5-2-8

(8) TA7357AP

The TA7357AP is used for sync-signal separation of the video signal in VCRs.

It is also used as the HD (horizontal drive) pulse generator; the pulse width becomes as narrow as TYP 2.5 μsec.

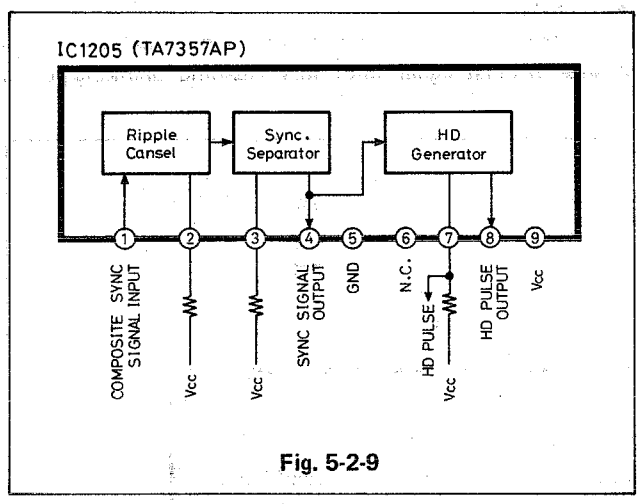


Fig. 5-2-9

(9) BA7007

The BA7007 is used to discriminate PAL/SECAM video signals in VCRs.

The BA7007 includes the following functions.

- Limiter amplifier
- Detector and slicer
- Tuning amplifier
- Comparator

The features of this IC are as follows

- High sensitivity for discrimination of PAL/SECAM signals.
- Stability in relation to fluctuations of the power supply voltage

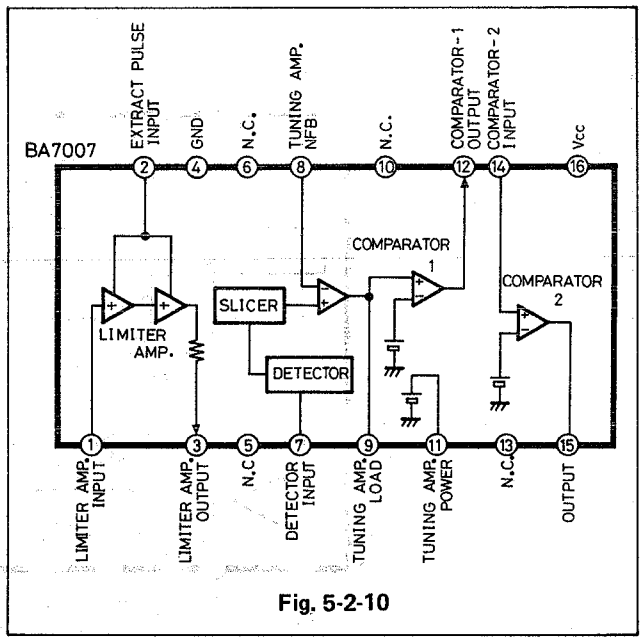


Fig. 5-2-10

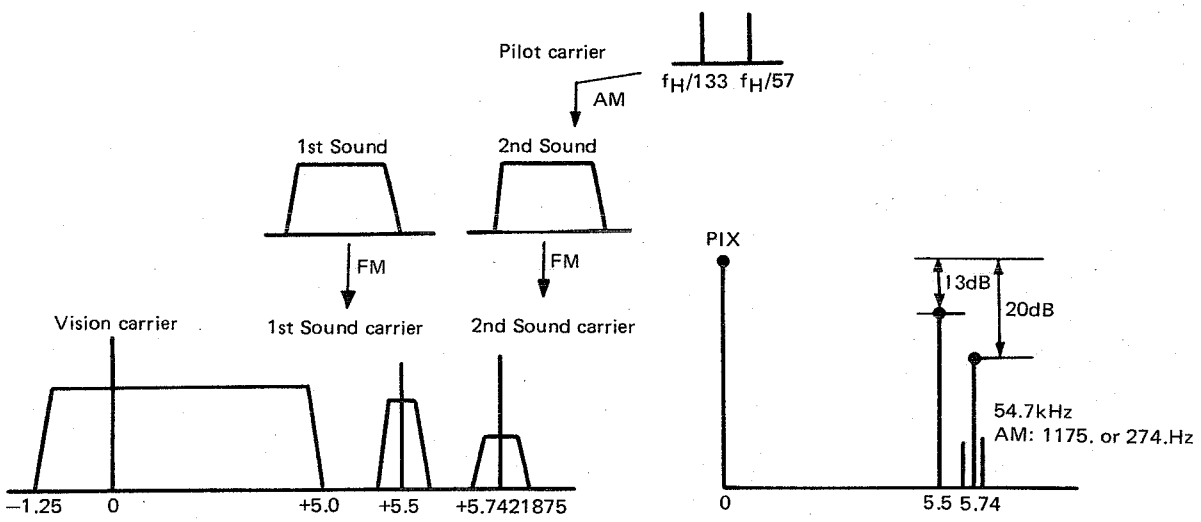
SUPPLEMENT

MULTIPLEX SOUND SYSTEM for VTC-M50
(WEST GERMANY)

MULTIPLEX SOUND SYSTEM

Transmission characteristics of the two-carrier system

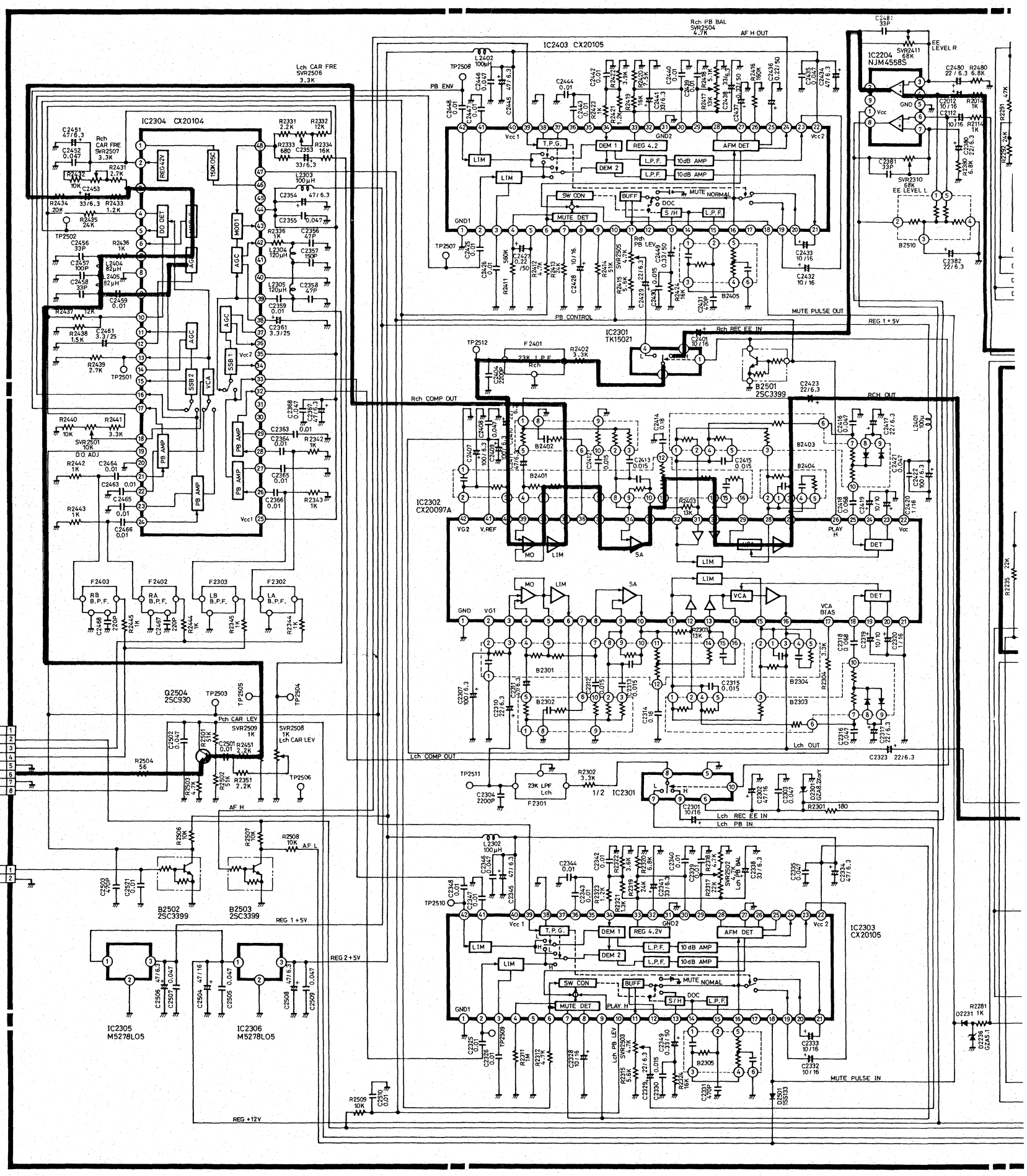
Characteristics	Television system B, G, H (PAL)	
	First sound carrier	Second sound carrier
Frequency referred to vision carrier (MHz)	5.5	5.472 (5.5 + 0.242)
Power referred to vision carrier (dB)	–13	–20
Modulation	FM	FM
Deviation (kHz)	±50	±50
Audio-frequency range (Hz)	40 to 15000	40 to 15000
Pre-emphasis (μs)	50	50
Mode		
Mono	Mono 1	Mono 1
Stereo	(L+R)/2	R
Dual Tone	M (Mono 1)	S (Mono 2)
Identification		
Pilot carrier frequency		54.6875kHz (3.5 x fH)
Modulation of pilot carrier		AM
Modulation depth		50%
Identification frequencies on the pilot carrier		
Mono		0Hz
Stereo		117.5Hz (fH/133)
Dual Tone		274.1Hz (fH/57)
Frequency deviation of the second sound carrier by the pilot carrier		±2.5kHz



AUDIO CIRCUI

REC MODE SI

P.C.B. AD-1 AUDIO CIRC

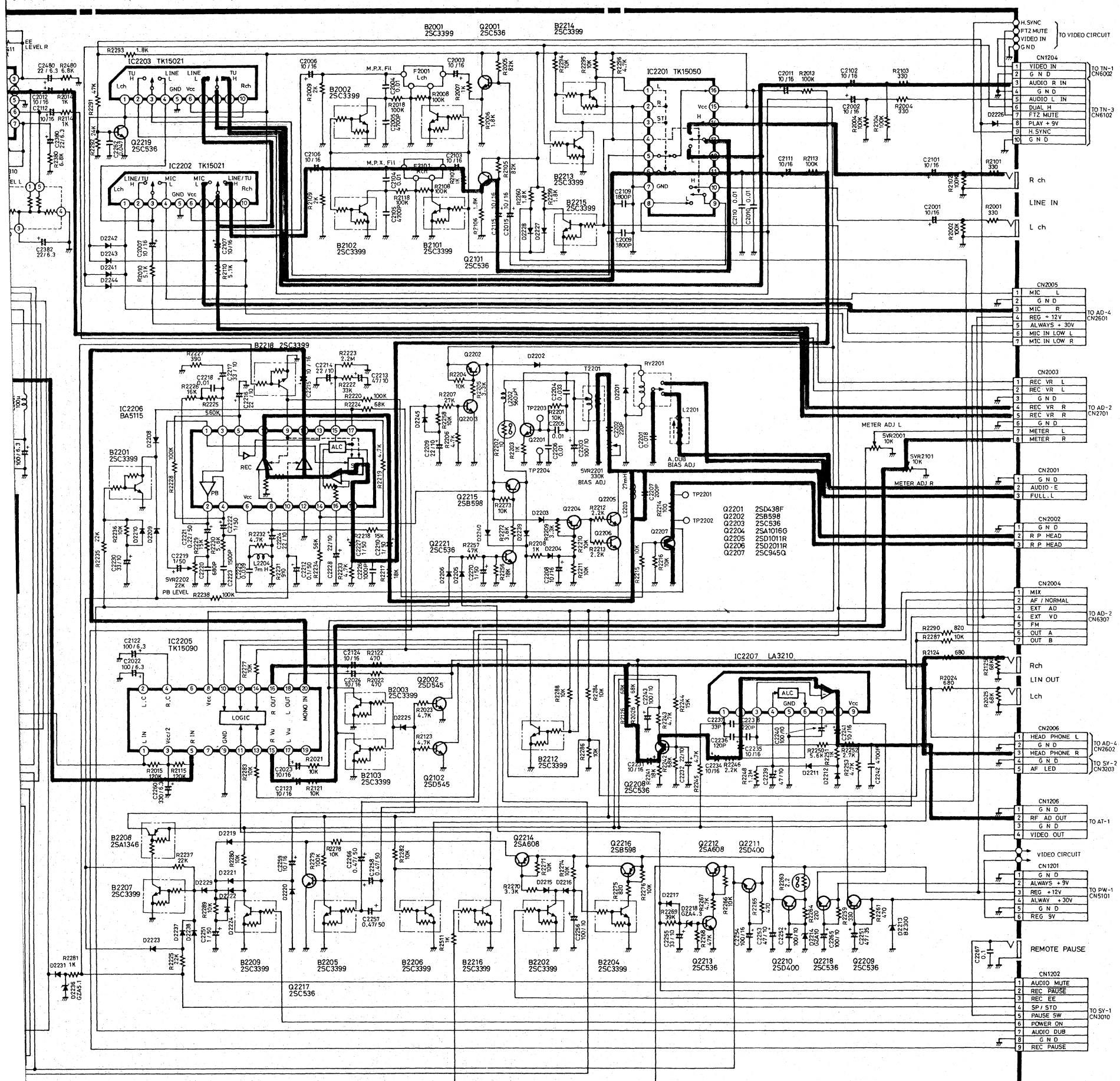


P.C.B. AD-3 AF PREAMP CIRCUIT (143-0-230V60821)

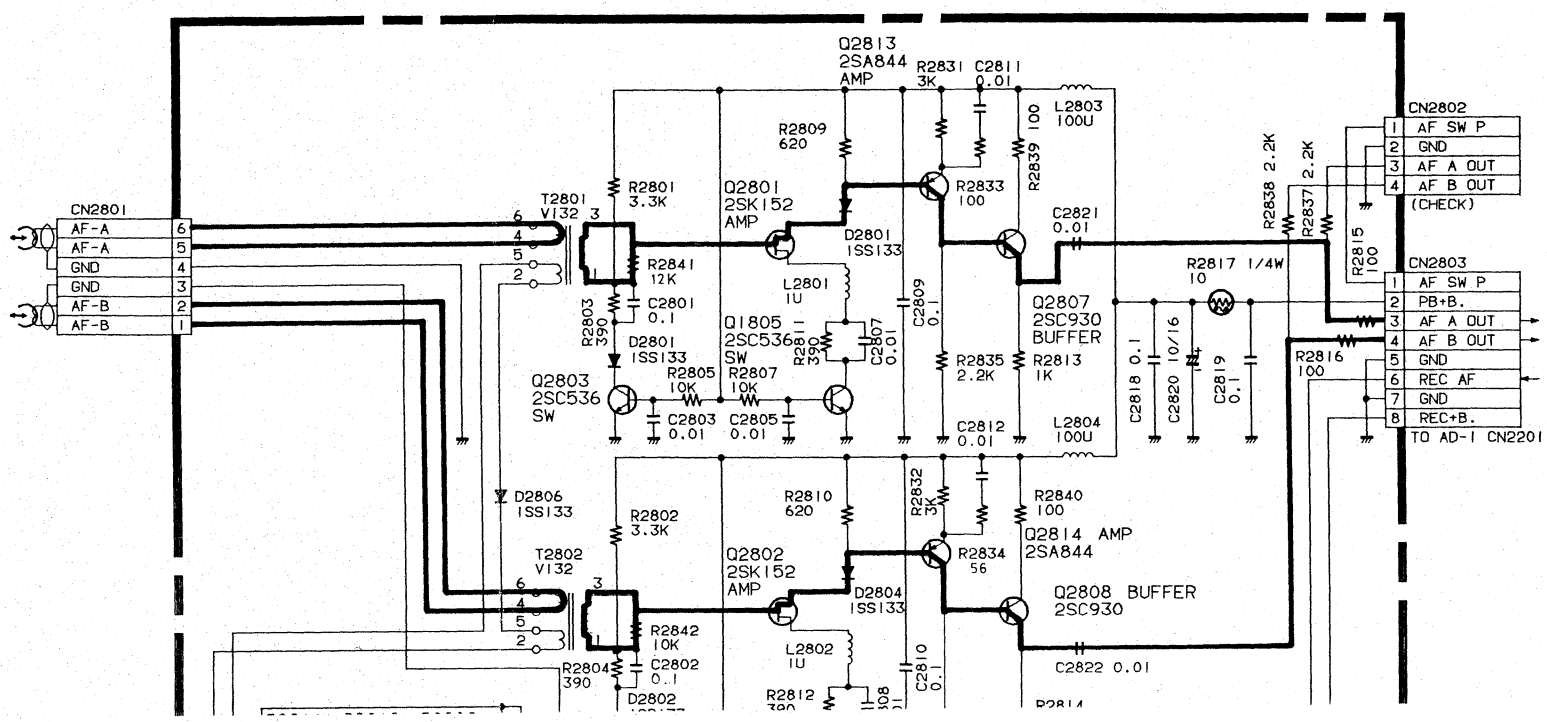
CIRCUIT DIAGRAM

ODE SIGNAL FLOW

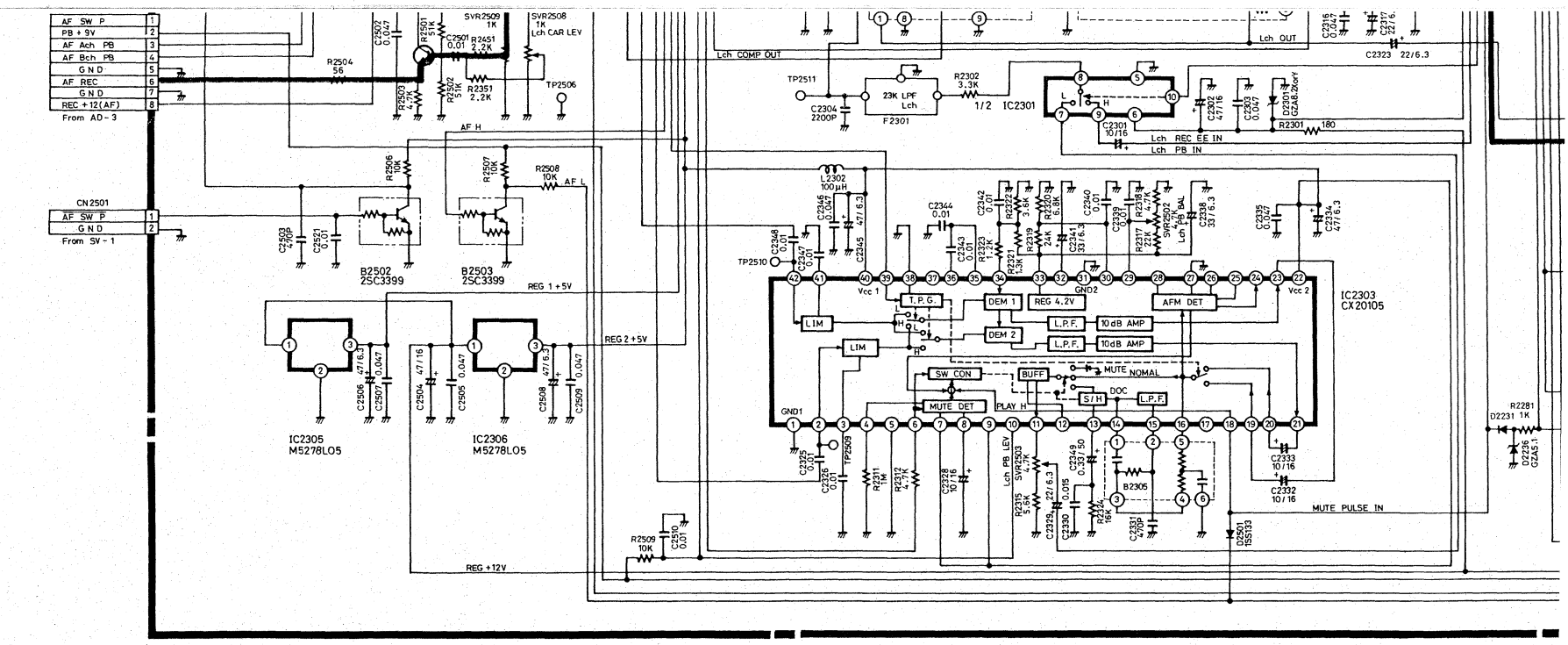
-1 AUDIO CIRCUIT (143-0-230V68100)

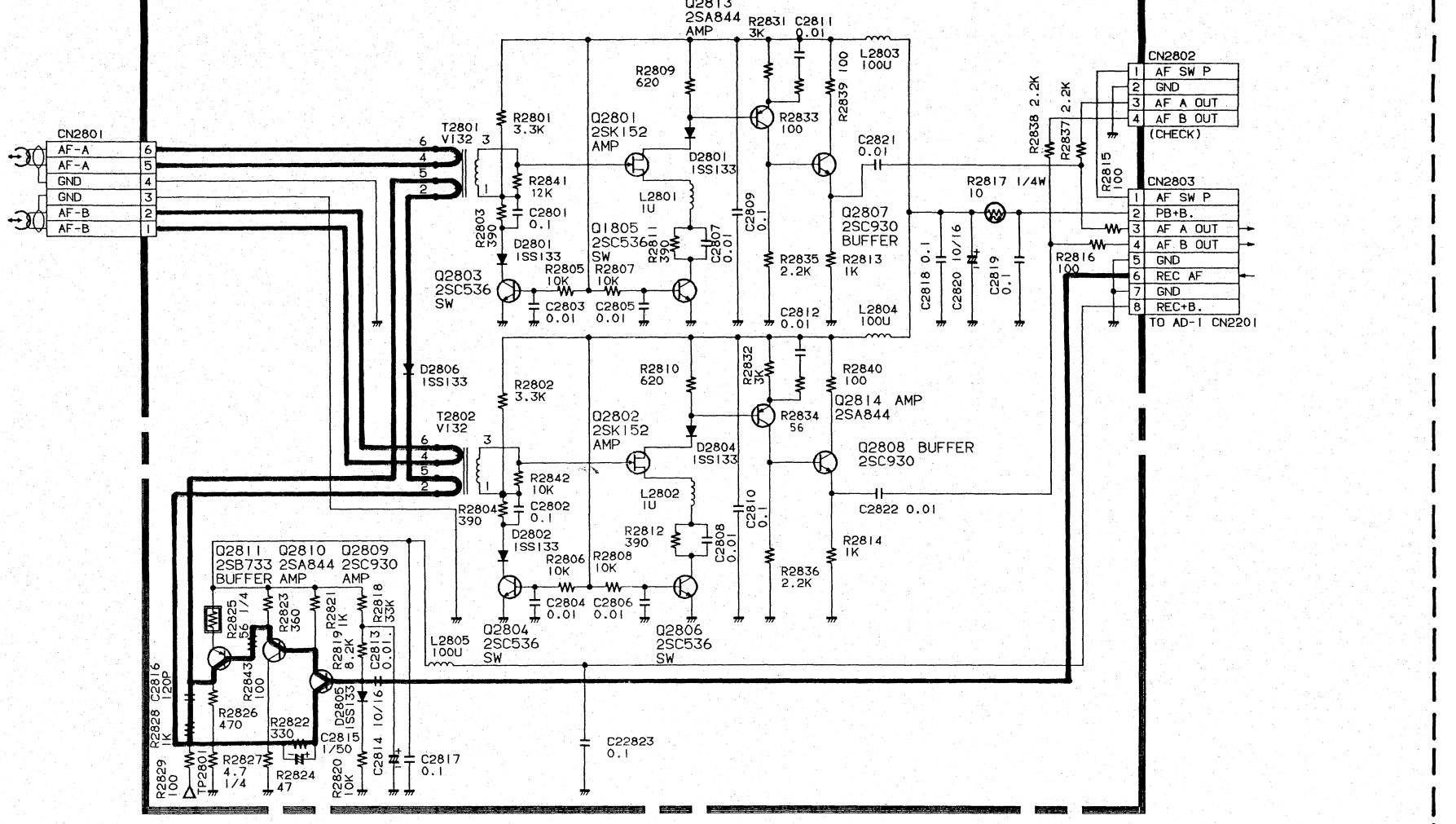


P.C.B. AD-3 AF PREAMP CIRCUIT (143-0-230V60821)

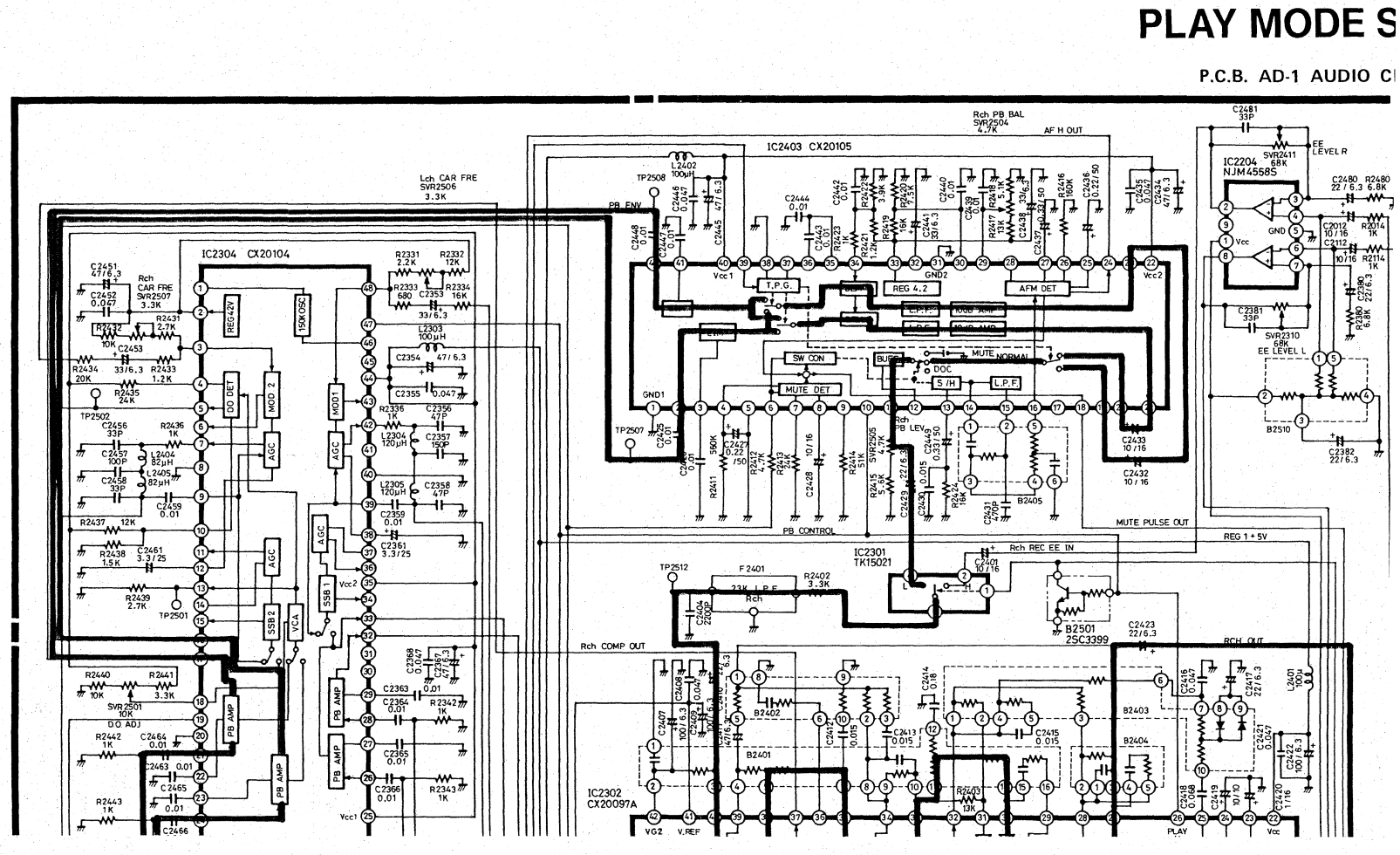


P.C.B. AD-3 AF PREAMP CIRCUIT (143-0-230V60821)





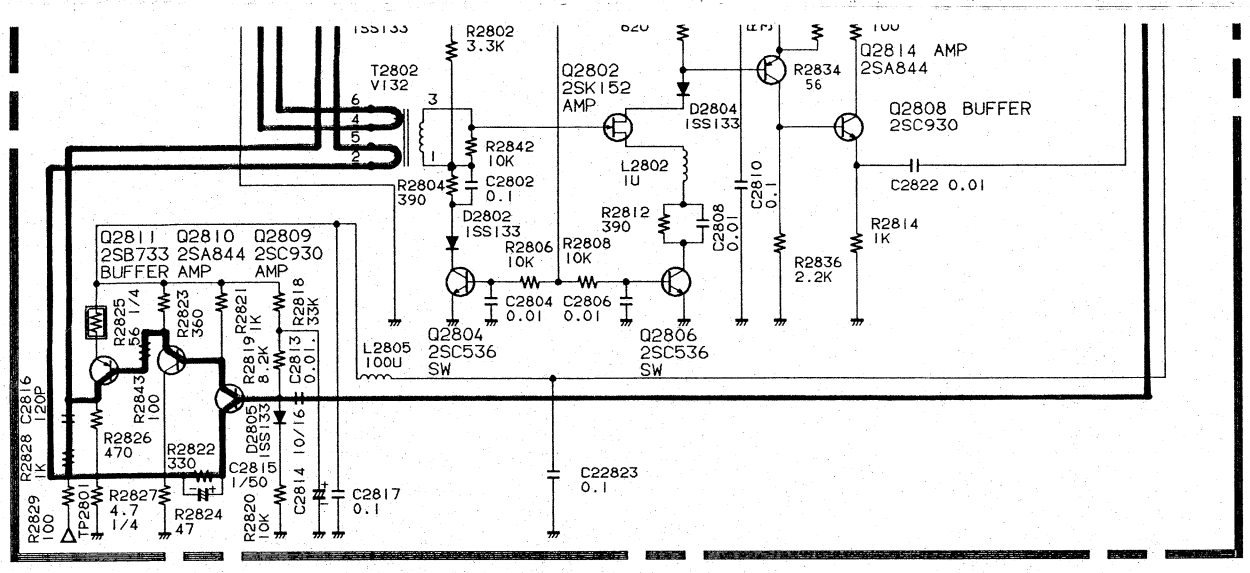
P.C.B. AD-1 AUDIO C



P.C.B. AD-3 AF PREAMP CIRCUIT (143-0-230V60821)

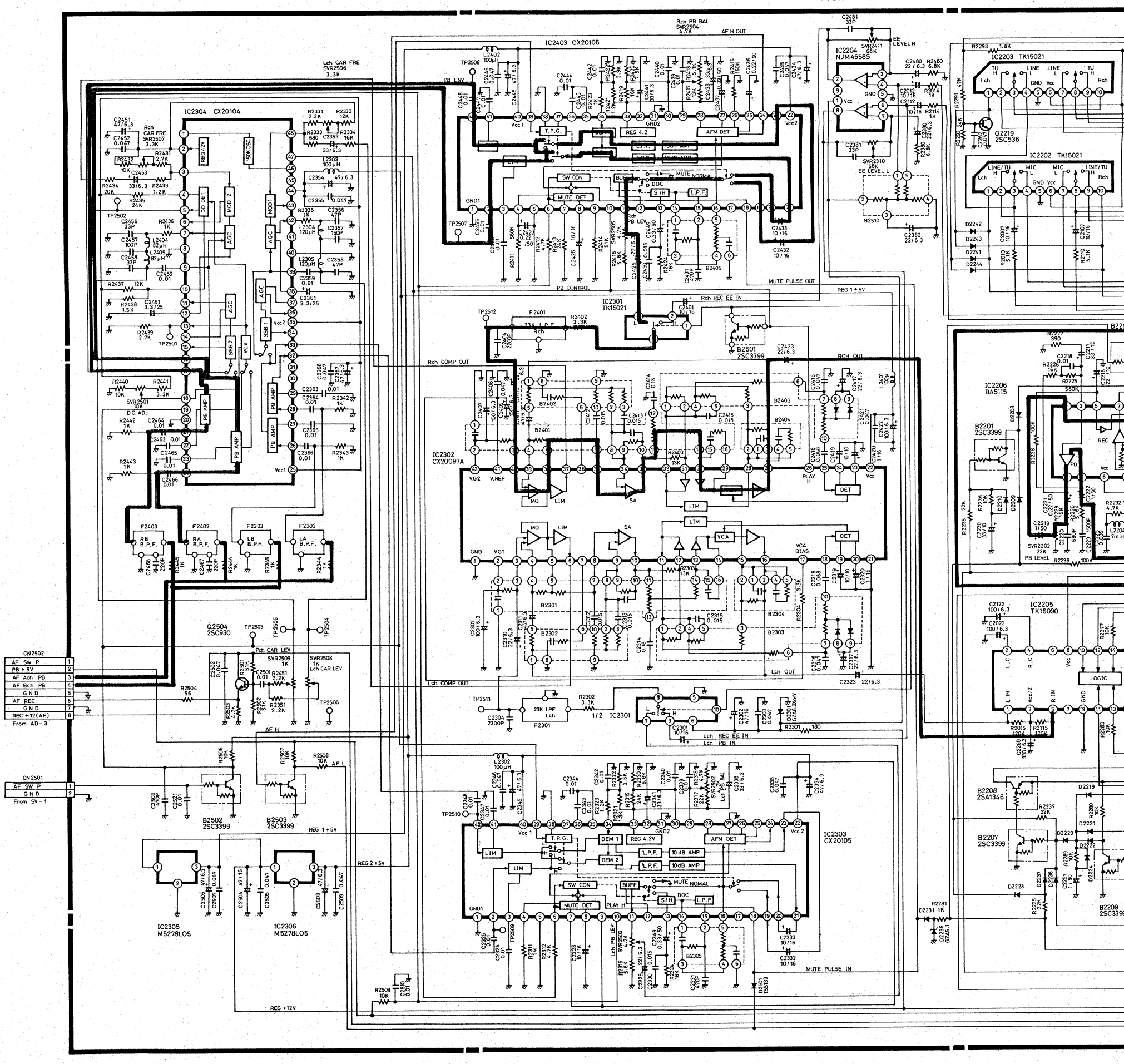
ODE SIGNAL FLOW

1 AUDIO CIRCUIT (143-0-230V68100)



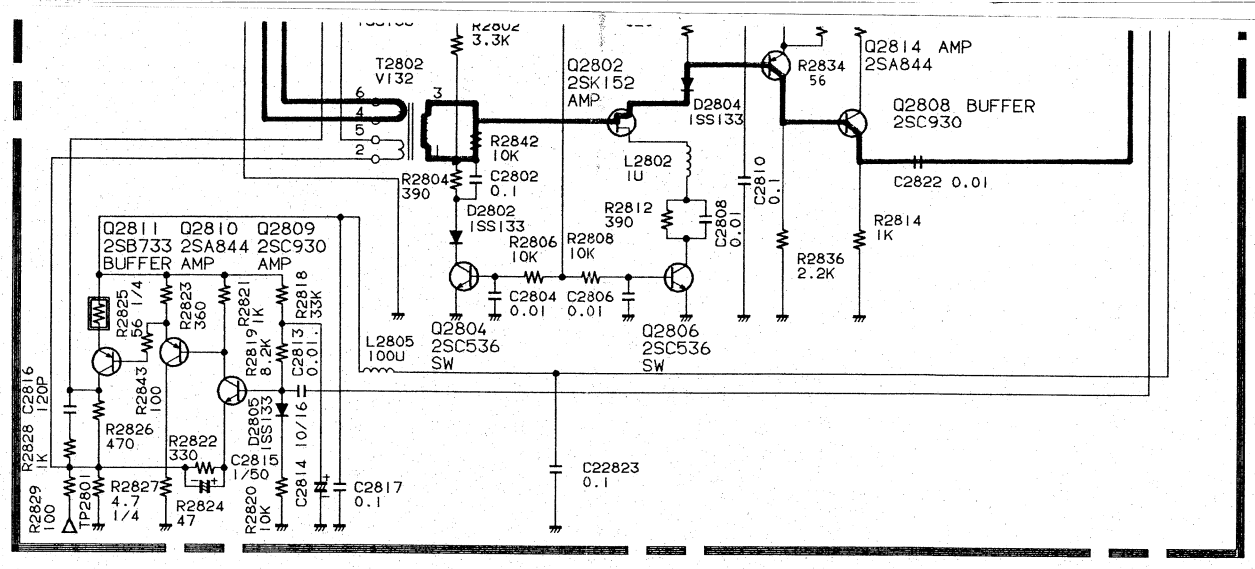
PLAY MODE SIGNAL FLOW

P.C.B. AD-1 AUDIO CIRCUIT (143-0-230V68100)



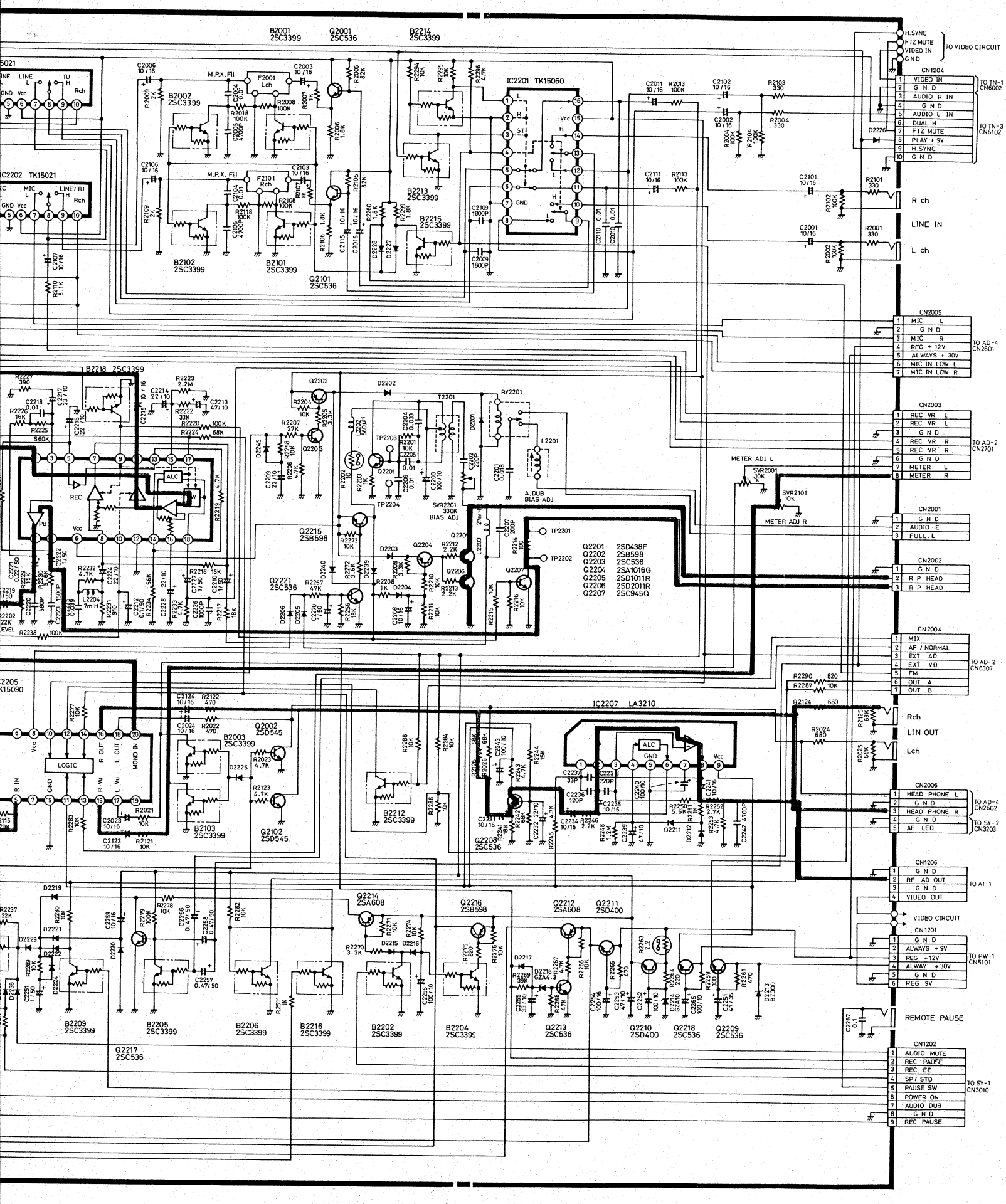
Classification of the color lines for REC/PLAY m

Green system		
NORMAL AUDIO and AUDIO FM (F/Rch) signal	AUDIO FM Lch signal	A



FLOW

230V68100)

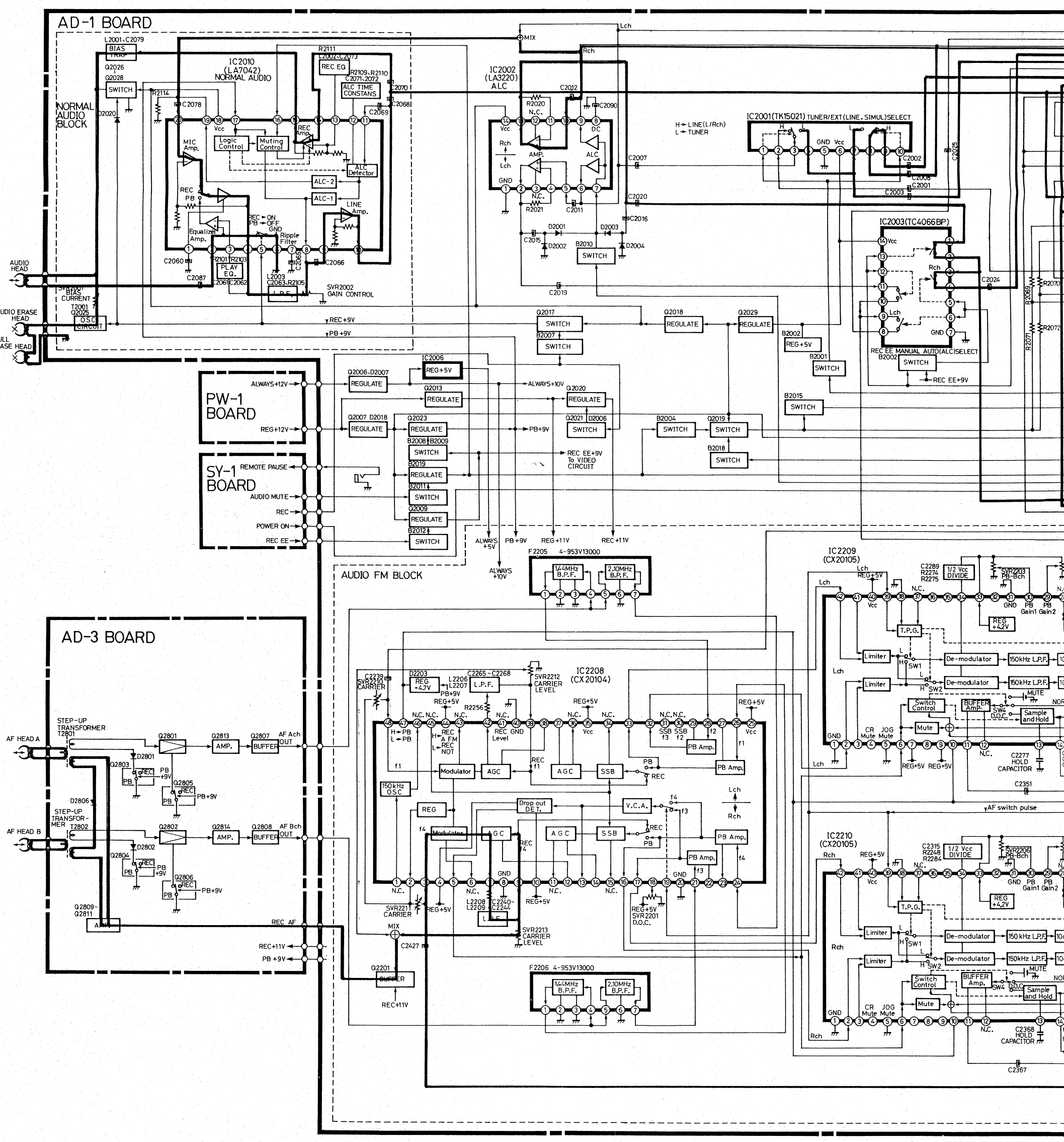


PLAY mode signal flow.

	Red system
nal	AUDIO FM Rch signal

AUDIO CIRCUIT BLO

REC MODE SIGN

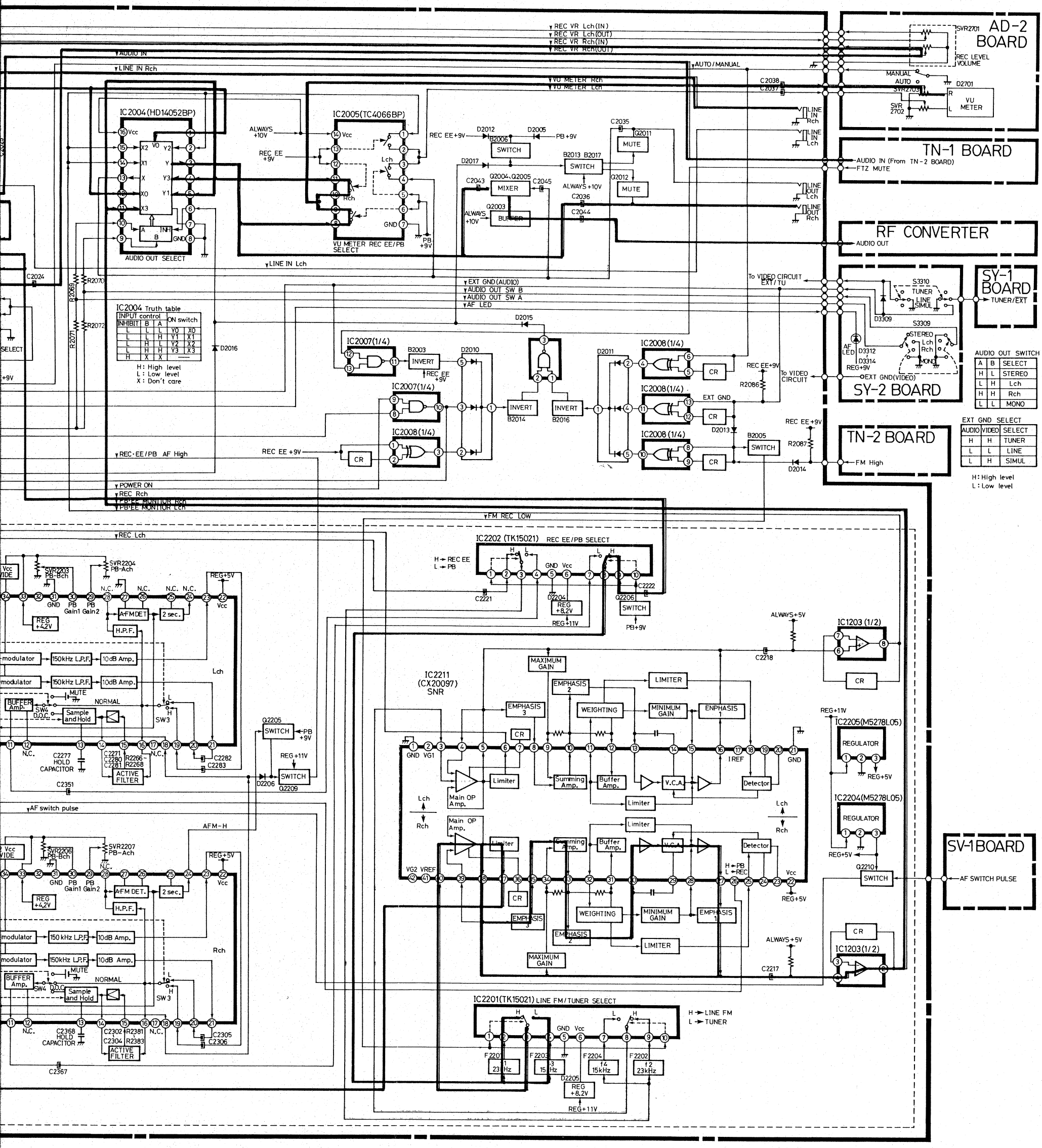


Classification of the color lines for RE

Green system	
NORMAL AUDIO and AUDIO FM (L/Rch) signal	AUDIO Lch (FM)

UIT BLOCK DIAGRAM

ODE SIGNAL FLOW

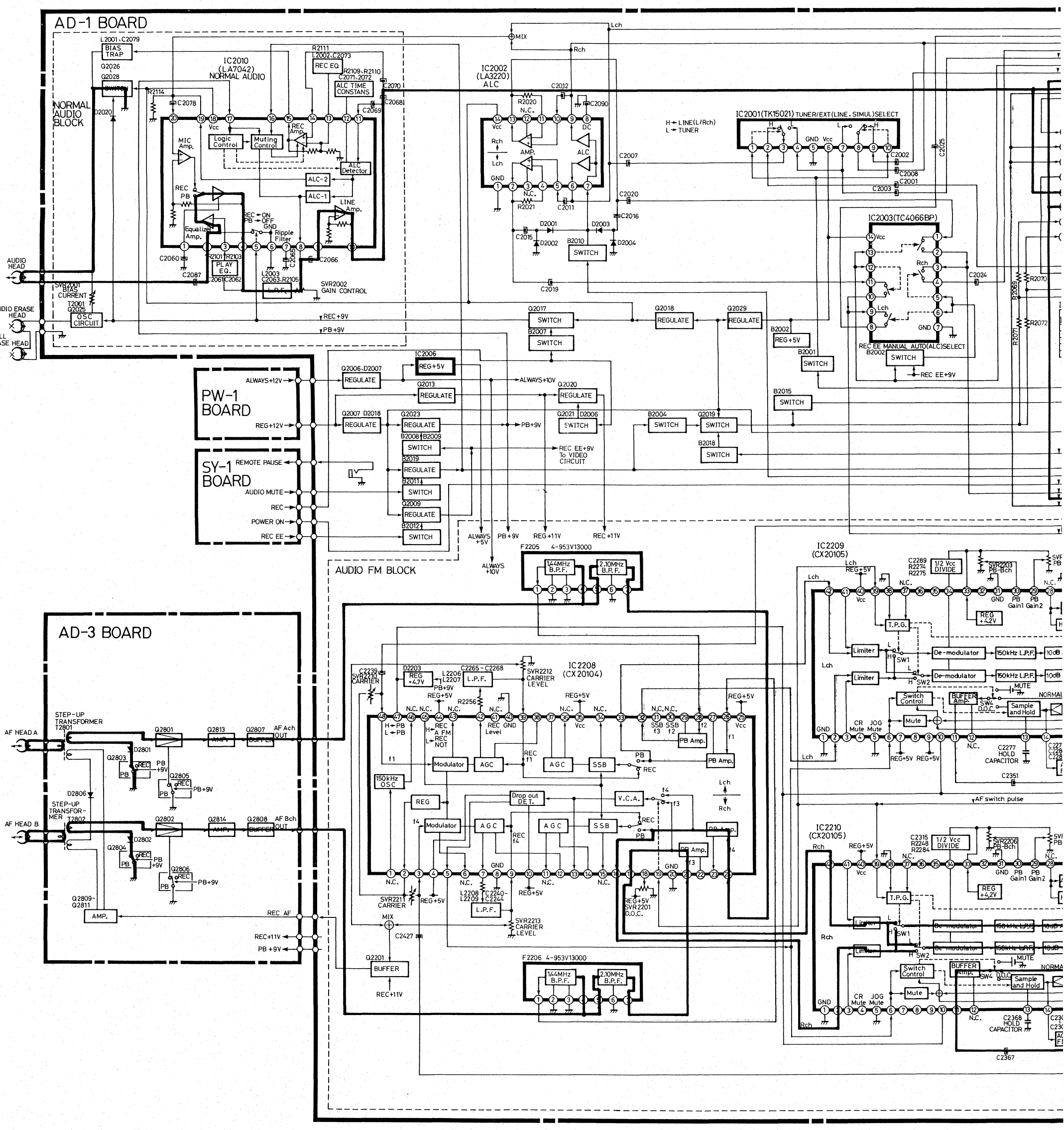


nes for REC/PLAY mode signal flow.

	Red system
D Lch (FM) signal	AUDIO Rch (FM) signal

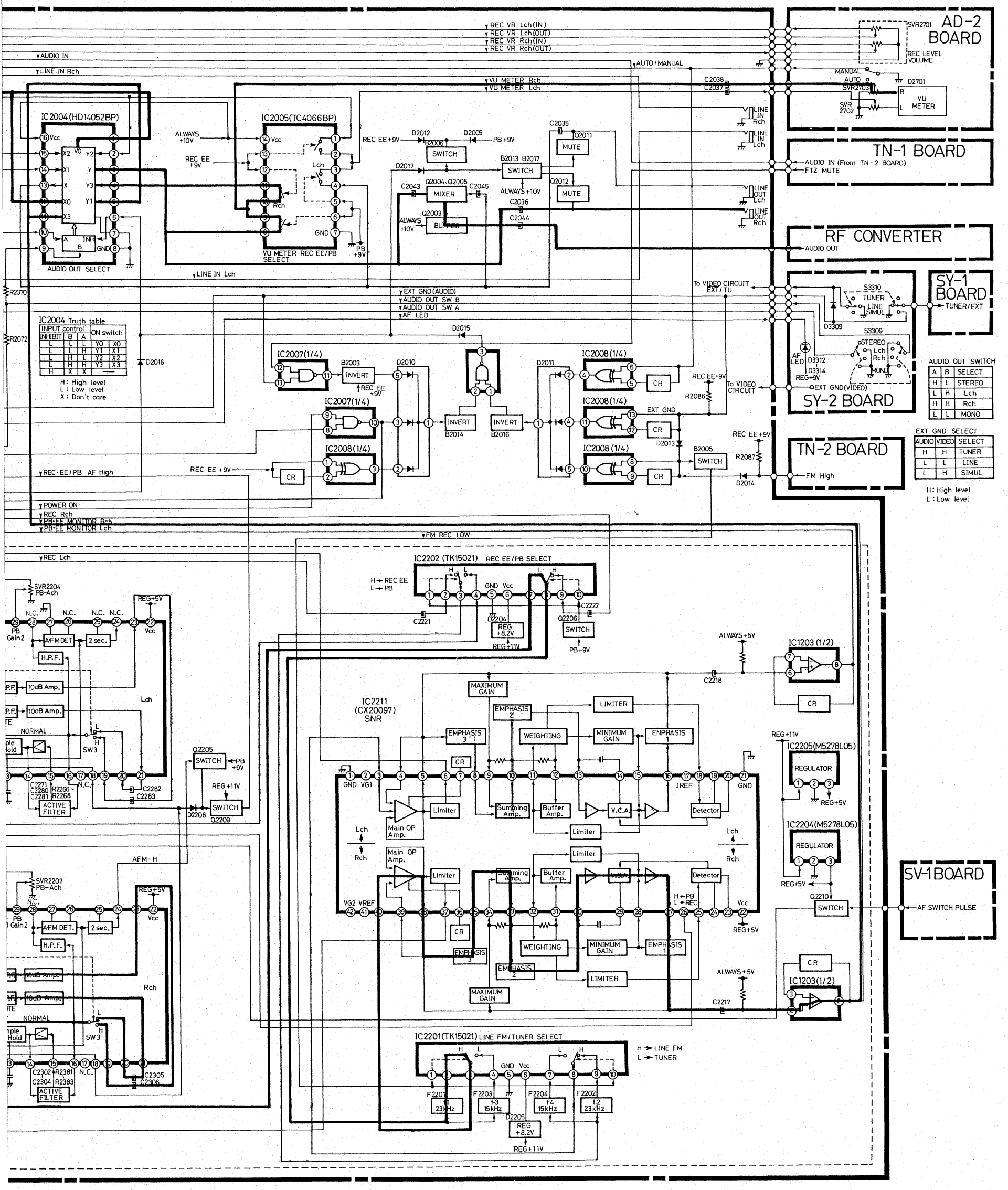
Green system	
NORMAL AUDIO and AUDIO FM (L/Rch) signal	AUDIO Lch (FM) s

PLAY MODE SIGNA


AD-1 BOARD
NORMAL AUDIO BLOCK
L2001、C2079
BIAS TRAP
Q2026
Q2028
SWITCH
D2020
R2114
IC2010 (LA7042) NORMAL AUDIO
R2111
L2002、C2073
REC EQ
R2109、R2110
C2071、2072
ALC TIME CONSTANS
C2070
C2068
C2069
C2078
MIC Amp.
Logic Control
Muting Control
REC Amp.
ALC Detector
ALC-2
ALC-1
REC PB
LINE Amp.
Equalize Amp.
REC→ON PB→OFF GND
Ripple Filter
C2060
C2087
R2101 R2103
PLAY EQ.
C2061 C2062
L2003 C2063 R2105
L.P.F.
C2066
C2065
SVR2002 GAIN CONTROL
AUDIO HEAD
SVR2001 BIAS CURRENT
T2001 Q2025
OSC CIRCUIT
AUDIO ERASE HEAD
FULL ERASE HEAD
REC+9V
PB+9V
MIX
Lch
Rch
IC2002 (LA3220) ALC
C2012
R2020 N.C.
C2090
C2007
C2020
C2016
R2021
C2011
D2001
D2003
D2002
D2004
C2015
B2010 SWITCH
C2019
H→LINE(L/Rch)
L→TUNER
IC2001(TK15021) TUNER/EXT(LINE.SIMUL)SELECT
C2002
C2008
C2001
C2003
C2025
IC2003(TC4066BP)
C2024
R2069
R2070
R2071
R2072
REC EE MANUAL AUTO(ALC)SELECT
B2002 SWITCH
REC EE+9V
Q2017 SWITCH
B2007 SWITCH
Q2018 REGULATE
Q2029 REGULATE
B2002 REG+5V
B2001 SWITCH
B2015 SWITCH
PW-1 BOARD
ALWAYS+12V
REG+12V
SY-1 BOARD
REMOTE PAUSE
AUDIO MUTE
REC
POWER ON
REC EE
IC2006 REG+5V
Q2006、D2007 REGULATE
Q2013 REGULATE
ALWAYS+10V
Q2020 REGULATE
Q2007 D2018 REGULATE
Q2023 REGULATE
PB+9V
Q2021 D2006 SWITCH
B2004 SWITCH
Q2019 SWITCH
B2018 SWITCH
B2008 B2009 SWITCH
REC EE+9V To VIDEO CIRCUIT
B2019 REGULATE
B2011 SWITCH
Q2009 REGULATE
B2012 SWITCH
ALWAYS +5V
PB+9V
REG+11V
REC+11V
ALWAYS +10V
F2205 4-953V13000
144MHz B.P.F.
2.10MHz B.P.F.
AUDIO FM BLOCK
AD-3 BOARD
STEP-UP TRANSFORMER T2801
AF HEAD A
Q2801
Q2813 AMP.
Q2807 BUFFER
AF Ach OUT
Q2803
D2801
REC
PB
PB +9V
Q2805
D2806
STEP-UP TRANSFOR-MER T2802
AF HEAD B
Q2802
Q2814 AMP.
Q2808 BUFFER
AF Bch OUT
D2802
Q2804
Q2806
Q2809~ Q2811 AMP.
REC AF
REC+11V
PB +9V
IC2208 (CX20104)
C2239 SVR2210 CARRIER
D2203 REG +4.7V
L2206 L2207 L.P.F.
C2265~C2268
SVR2212 CARRIER LEVEL
PB+9V
REG+5V
R2256
N.C. N.C. N.C.

	Red system
M) signal	AUDIO Rch (FM) signal

GNAL FLOW

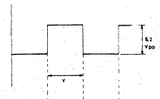


VIDEO CIRCUIT BLO

PLAY MODE SIGN

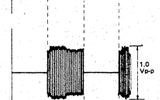
PLAY MODE MAIN SIGNAL	
——— Video (Y/C) signal	
Luminance (Y) signal	
——— Chrominance (C) signal	

RF SWITCH PULSE
CN1504 (3)



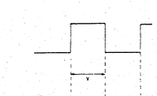
ENVELOPE WAVE
CN1504 (1)

RF B channel
IC1501 (16)

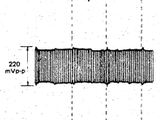


RF A channel
IC1501 (17)

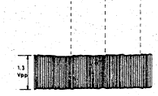
RF SWITCH PULSE
CN1504 (3)



D.O.C. 1HDL
P1502 (7)



PB-Y signal
P1502 (10)



Continuous A channel and B channel video signals are obtained.

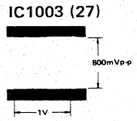
Function of both the comb-type noise canceller and the delay line to produce the signal 1H prior to drop-out.

The high-range components (unnecessary FM carrier component, etc.) are eliminated, and returned to the original Y signal component which includes the DC component.

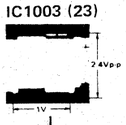
The Y-FM signal is de-modulated and then returned to the original Y signal.

$\pi/2$ phase shift of 4.43-M countinuous wave signal.

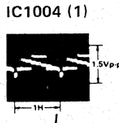
IC1003 (27)



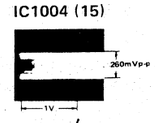
IC1003 (23)



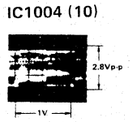
IC1004 (1)



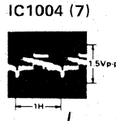
IC1004 (15)



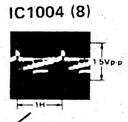
IC1004 (10)



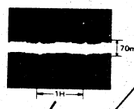
IC1004 (7)



IC1004 (8)



TP1008



VP-1 BOARD

VD-1 BOARD

IC1002 (LA7020)

IC1004 (LA7013)

CN1504 (3)

VIDEO HEAD A

VIDEO HEAD B

STEP-UP TRANS T1501

STEP-UP TRANS T1502

A channel

B channel

D1501

D1503

D1507

D1508

Q1501

Q1502

Q1503

Q1504

Q1506

REC

PB

PB+9V

MIXER

R1567

R1568

Q1516, Q1517

Q1513

Q1514, Q1515

SVR 1506

SVR 1505

Y·REC LEVEL

C·REC LEVEL

REC·C

REC·Y

Composite Video signal

Emitter follower

Q1009

BUFFER

DL1001

DELAY

Y signal

Q1008

VIDEO IN

REC EE+9V

SVR1003 CARRIER SETTING

SVR1002 EE LEVEL

SVR1001 COMPRESSOR

L1001 T1001 T1002

4.43MHz TRAP

L1003 C1027

TRAP

L1017 C1077 C1019 C1060

L.P.F.

Q1021

AMP.

NOISE CAN-CEL

SVR1012

C1081

PB+9V

De-modulator

ADD

Limiter

Vcc

GND

CR

N.C.

Sync. Separator

Clamp

Amp.

AGC Amp.

Non Linear Emphasis

FM Modulator

Emphasis Amp.

B/W Clip

Sync. AGC

Peak AGC

Q1020

BUFFER

C1089

Q1088

Q1087

Q1086

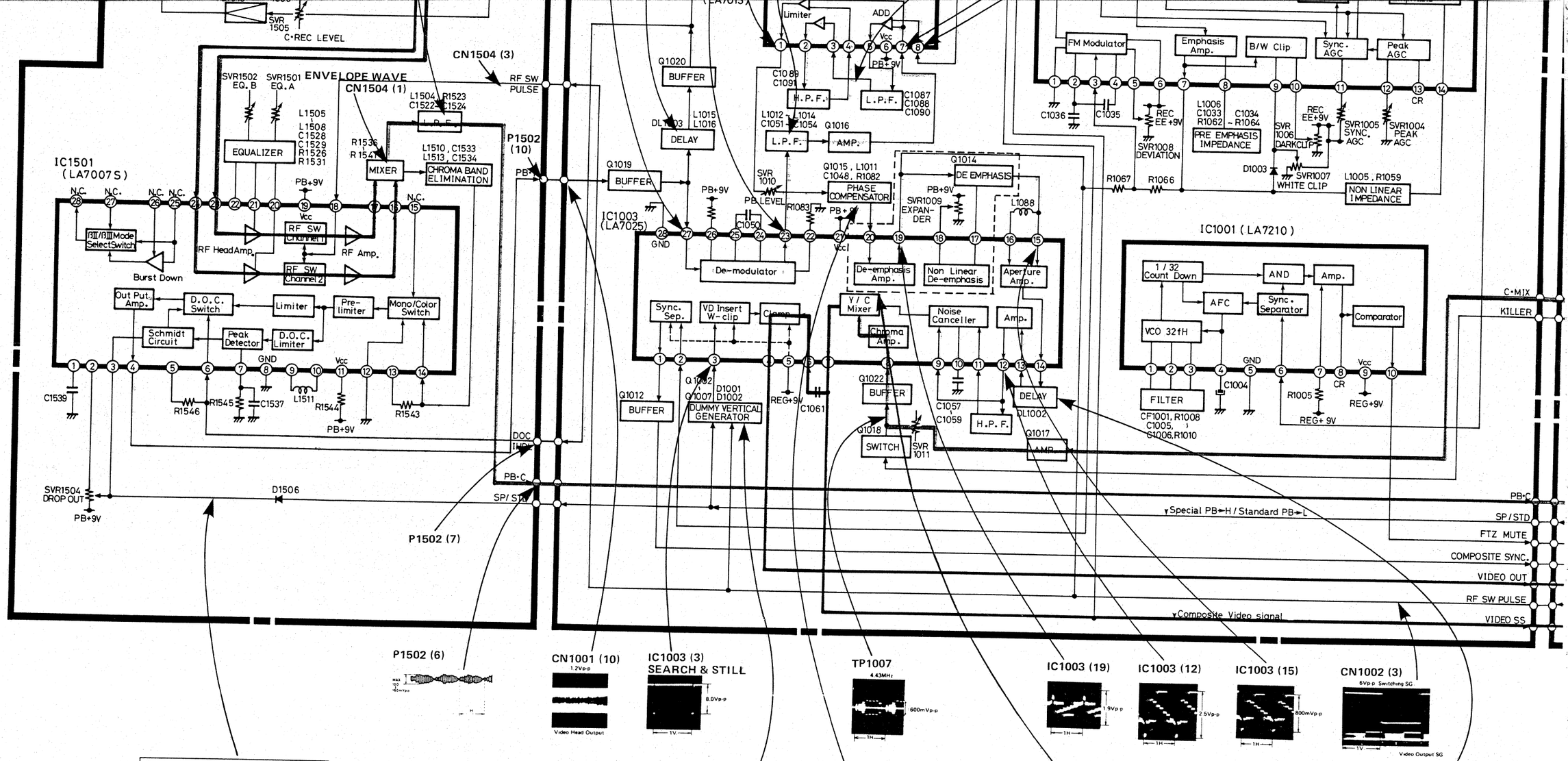
Q1085

Q1084

Q1083

Q1082

Q1081



P1502 (6)

Switches OFF D.O.C. circuit during special playback models.

CN1001 (10)

IC1003 (3)
SEARCH & STILL

Produces dummy vertical synchronizing signal during special playback modes, and adds it to video signal.

TP1007

The phase of the Y signal (processed by the playback circuit of the low-pass filter and de-modulated from the playback head) is corrected.

IC1003 (19)

IC1003 (12)

IC1003 (15)

The high-range component pre-emphasized during recording is returned to the original. Then non-linear de-emphasis (expansion) and wave-formation occur.

CN1002 (3)

C signal is delayed relative to Y signal because chrominance signal passes through narrow-band filter; Y and C times are coordinated by delaying the Y signal.
(Y signal delay: 600 nsec.)

Dur
supe
nan

JIT BLOCK DIAGRAM

ODE SIGNAL FLOW

-	CONTROL SIGNAL
	APC loop
nal	ACC loop
ignal	AFC loop (Special PB)
	APC-ID loop

se shift of 4.43-MHz ious wave signal.

Sets phase to +135° and −135° relative to chrominance signal of re-inserted burst.

Produces the flag for burst insertion.

No operation as comb filter during dummy SECAM.

Amplifier to cover attenuation by comb filter.

Period of pilot burst is not passed through the comb filter.

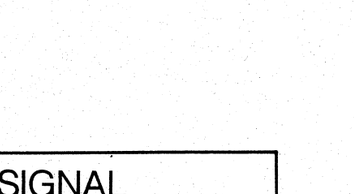
Determines time response of APC following APC loop.

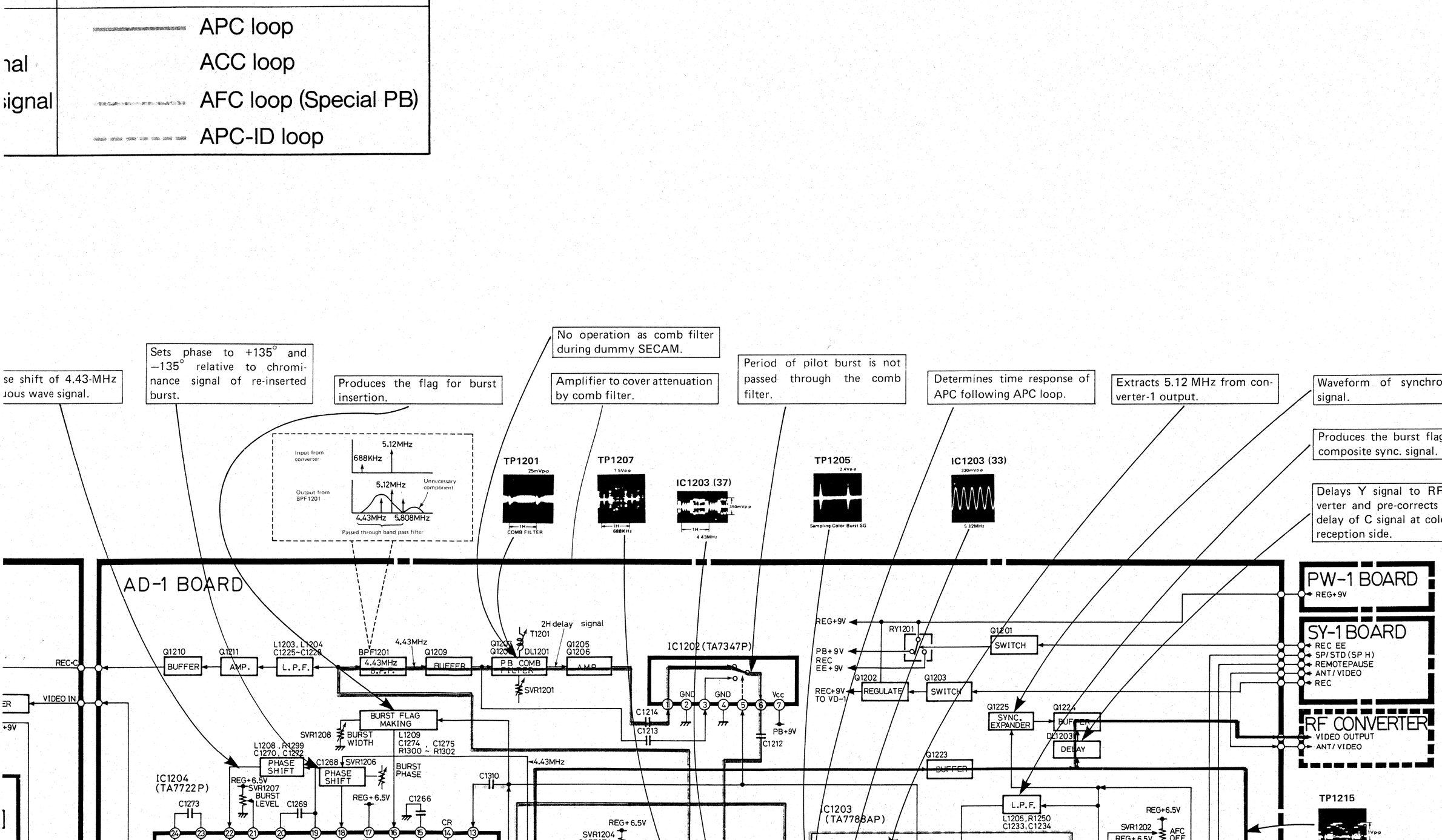
Extracts 5.12 MHz from converter-1 output.

Waveform of synchronizing signal.

Produces the burst flag from composite sync. signal.

Delays Y signal to RF converter and pre-corrects phase delay of C signal at color TV reception side.





VIDEO IN
+9V
BURST FLAG MAKING
SVR1208 BURST WIDTH
L1209 C1274 . C1275 R1300 ~ R1302
L1208 . R1299 C1270 . C1292
PHASE SHIFT
C1268
SVR1206 PHASE SHIFT
BURST PHASE
4.43MHz
C1310
IC1204 (TA7722P)
REG+6.5V
SVR1207 BURST LEVEL
C1269
C1266
C1273
Phase Invert
Gain Control Amp
Burst Gate Generation
Pulse Wavetorm
Jump DET.
Buffer
F.F.
Switch Control
Cleaning Flag Gen.
DL Drive
Gain ControlAmp
Switch
VCO
AFC DET.
fH OSC
DELAY
DL1202
T1202 HORIZON -TAL PHASE
C1286
SVR1211 1H GAIN
SVR 1210
C1283 SPECIAL PB VCO
C1280
AFC FILTER
R1304 , C1281, C1282
C1214
C1213
C1212
PB+9V
Q1225 SYNC. EXPANDER
Q1224 BUFFER
D1203
DELAY
Q1223 BUFFER
L.P.F.
L1205, R1250 C1233, C1234
RF CONVERTER
VIDEO OUTPUT
ANT/ VIDEO
TP1215
1Vp-p
1H
VIDEO LINE OUT
REMOTE PAUSE
VIDEO LINE IN
From AUDIO hi-fi CIRCUIT
CIRCUIT EXT/TU
REC EE+9V
B1201 SWITCH
TUNER AUDIO MUTE SIGNAL
9V
EXT+SIMUL TUNER
0V
D1206
D1201
TN-2 BOARD
TUNER AUDIO MUTE
TN-1 BOARD
VIDEO IN
FTZ MUTE
SV-1 BOARD
COMPOSITE SYNC.
RF SWITCH PULSE
D1202
C1203 (TA7788AP)
SVR1204 CARRIEA LEAK
C1245
R1261
R1260 C1242 ACK FILTER
APC FILTER
R1257 C1241
CF1201 5.12MHz FILTER
R1252 C1236, C1237 AFC FILTER
SVR1202 AFC OFF SET
Q1214 SWITCH
SVR1203 VCO FREE OSC. FREQUENCY
Chroma mode
REC-Killer
CONV.2
NTSC
PAL
Burst Flag1
Burst Emphasis
ACK
Killer
Killer DET.
Chroma Inverter
F.F.
Amp.
PB
REC
Burst ID DET.
Phase Invert
RF SW
1/2fH
Burst Flag 2
PB-PAL
Pilot Burst Flag
BurstFlag1
Burst Flag 2
VCO
AFC DET.
HD GEN.
REC
PB
5fH
CONV.1
Counter
1/2
1/2fH
APC-ID
(44±1/8)fH
NTSC
PAL
NTSC 3.58
Approximate 5.5MHz
Burst Flag2
SYNC. DET.
4.43MHz
Reference signal
Pilot Burst Flag
REC+PAL
Pilot Burst ADD
ACC Amp.
Channel Hold
fsc VXO
3fsc OSC.
ACC DET.
PB or SECAM
C1263
R1292
SVR1205 PILOT BURST
REG+ 6.5V
R1286 C1256 C1254 C1258
ACC FILTER
C1253
PB+9V
R1248
C1249 R1277
R1247
PHASE SHIFT
R1273
N.C.
Vcc
GND
GND2
Vcc2
IC1205 (TA7357AP)
Ripple Cancel
Sync. Separator
HD Generator
Q1221 MODE SWITCH
Q1222 BUFFER
R1317
R1314
R1314
REG+ 9V
REG+9V
REG+9V
Q1216 SWITCH
BPF1202 4.43MHz B.P.F.
Q1218 BUFFER
Q1220 ACC FILTER SWITCH
X1201 C1250 R1348 C1251
VC1201 VXO
VXO FREE REFERENCE
IC1201 (LA7016)
Q1215 BUFFER
Q1204 BUFFER
REC EE+9V
SP H
Composite Video signal
FTZ mute signal
Composite sync.(V and H) signal
Composite sync.
C-MIX
KILLER
C signal + DC level
PB-C
SP/STD
FTZ MUTE
COMPOSITE SYNC.
VIDEO OUT
RF SW PULSE
VIDEO SS
IC1203 (1)
IC1203 (30)
4.43MHz
1Vp-p
3.5V
1H
IC1203 (7)
IC1203 (28
688KHz
150mVp-p
4Vp-p
1H
TP1209
680mVp-p
4.43MHz
IC1203 (14)
600mVp-p
4.43MHz

e to ance row- s are the)

During color, DC bias is superimposed on the chrominance signal and output.

Delays chrominance signal by 1H during special playback modes.

Separates the synchronizing signal from the composite video signal.

Switches IC1204 pin 3 voltage: LOW level during normal playback, constant fixed level during special playback.

Determines time response of AFC following AFC loop.

Increases ACC response speed during special playback.

Determines time response of ACC following ACC loop.

During playback becomes X'tal OSC. of reference frequency (f_{sc}); during recording becomes VXO and locks to sub-carrier frequency of input.

Improves time response of AFC and APC during switching by the RF switch pulse.

$-\pi/4$ phase shifter.

VIDEO CIRCUIT BLO

REC MODE SIGNA

REC MODE MAINSIGNAL	
——— Video (Y/C) signal	
Luminance (Y) signal	
——— Chrominance (C) signal	

The C signal is superimposed on the Y signal.

Chrominance band width trap.

Emphasis of characteristic non-fluctuation (by level).

Y signal is pre-delayed in order to coincide Y signal with delay by color-processing circuit.

Passes signal

CN1001 (1)

820mVp-p 3.85MHz

NON FM MODULATE

820mVp-p

FM MODULATED

CN1001 (2)

688KHz

600mVp-p

TP1005

TP1002

800mVp-p

IC1002 (18)

2.1Vp-p

IC1002 (16)

480mVp-p

TP1010

1Vp-p

Video INPUT

VP-1 BOARD

VD-1 BOARD

STEP-UP TRANS T1501

VIDEO HEAD A

VIDEO HEAD B

STEP-UP TRANS T1502

D1501

D1503

D1508

D1507

▼A channel

▼B channel

Q1501

Q1502

Q1503

Q1504

Q1506

REC

PB

PB+9V

MIXER

R1567 R1568

Y•REC LEVEL

Q1516, Q1517

Q1514, Q1515

Q1513

SVR 1506

SVR 1505

C•REC LEVEL

SVR1502 SVR1501

REC•C

REC•Y

RF SW

Q1020 BUFFER

IC1004 (LA7013)

L1017 ~ L1019

C1077 ~ C1080

Q1021

L.P. F.

AMP.

NOISE CAN-SEL SVR1012

C1081

De-modulator

ADD

Limiter

C1089 C1091

IC1002 (LA7020)

▼Composite Video signal

L1003 C1027

TRAP

Emitter follower

Q1009

BUFFER

REC EE•9V

SVR1003 CARRIER SETTING

SVR1002 EE LEVEL

DL1001

DELAY

L1001 T1001 T1002

4.43MHz TRAP

▼Y signal

Q1008

BUFFER

VIDEO IN

SVR1001 COMPRESSOR

GND

Vcc

N.C.

CR

Sync. Separator

Clamp

Amp.

AGC Amp.

Non Linear Emphasis

FM Modulator

Emphasis Amp.

B/W Clip

Sync. AGC

Peak AGC

MIXER
R1567
R1568
Y•REC LEVEL
Q1516, Q1517
Q1514, Q1515
Q1513
SVR 1506
SVR 1505
C•REC LEVEL

IC1501
(LA7007S)
N.C.
(II/3)III Mode Select Switch
Burst Down
RF HeadAmp.
RF SW Channel 1
RF SW Channel 2
RF Amp.
Vcc
PB+9V
Out Put Amp.
D.O.C. Switch
Limiter
Pre-limiter
Mono/Color Switch
Schmidt Circuit
Peak Detector
D.O.C. Limiter
GND
C1539
R1546
R1545
C1537
L1511
R1544
R1543
PB+9V

SVR1502 EQ. B
SVR1501 EQ. A
EQUALIZER
L1505
L1508
C1528
C1529
C1526
R1531
R1536
R1541
MIXER
L1504, R1523
C1522~ C1524
L.P.F.
L1510, C1533
L1513, C1534
CHROMA BAND ELIMINATION

SVR1504 DROP OUT
PB+9V
D1506

RF SW PULSE
PB•Y
DOC
IHDL
PB•C
SP/STD

IC1004
(LA7013)
C1081
PB+9V
NOISE CAN-CEL
SVR1012
GND
De-modulator
Vcc
ADD
Limiter
ADD
Vcc
PB+9V
C1089
C1091
H.P.F.
L.P.F.
C1087
C1088
C1090
Q1020
BUFFER
DL1003
L1015
L1016
DELAY
L1012 ~ L1014
C1051 ~ C1054
L.P.F.
Q1016
AMP.
Q1019
BUFFER
PB+9V
SVR 1010
PB LEVEL
Q1015, L1011
C1048, R1082
PHASE COMPENSATOR
R1083
PB+9V
C1050
Q1014
DE EMPHASIS
PB+9V
SVR1009 EXPAN-DER
L1088

IC1003
(LA7025)
GND
Vcc
De-modulator
De-emphasis Amp.
Non Linear De-emphasis
Aperture Amp.
Sync. Sep.
VD Insert W-clip
Clamp
Y/C Mixer
Chroma Amp.
Noise Canceller
Amp.
Q1012
BUFFER
Q1002
Q1007
D1001
D1002
DUMMY VERTICAL GENERATOR
REG+9V
C1061
Q1022
BUFFER
C1057
C1059
DELAY
DL1002
H.P.F.
Q1018
SWITCH
SVR 1011
Q1017
AMP.

CR
GND
Vcc
N.C.
Sync. Separator
Clamp
Amp.
AGC Amp.
Non Linear Emphasis
FM Modulator
Emphasis Amp.
B/W Clip
Sync. AGC
Peak AGC
C1036
C1035
REC EE+9V
SVR1008 DEVIATION
L1006
C1033
R1062
C1034
~ R1064
PRE EMPHASIS IMPEDANCE
SVR 1006 DARKCLIP
REC EE+9V
SVR1005 SYNC. AGC
SVR1004 PEAK AGC
D1003
SVR1007 WHITE CLIP
L1005, R1059
NON LINEAR IMPEDANCE
R1067
R1066

IC1001 (LA7210)
1/32 Count Down
AND
Amp.
AFC
Sync. Separator
Comparator
VCO 32fH
GND
Vcc
C1004
CR
R1005
REG+9V
FILTER
CF1001, R1008
C1005, C1006, R1010

C•MIX
KILLER
PB•C
SP/STD
FTZ MUTE
COMPOSITE SYNC.
VIDEO OUT
RF SW PULSE
VIDEO SS

▼Special PB→H / Standard PB→L
▼Composite Video signal

TP1007
4.43MHz
1H

CN1002 (3)
6Vp-p Switching SG
1V
Video Output SG

Detects TV broadcast signal; changes FTZ signal to High level when there is no TV broadcast.

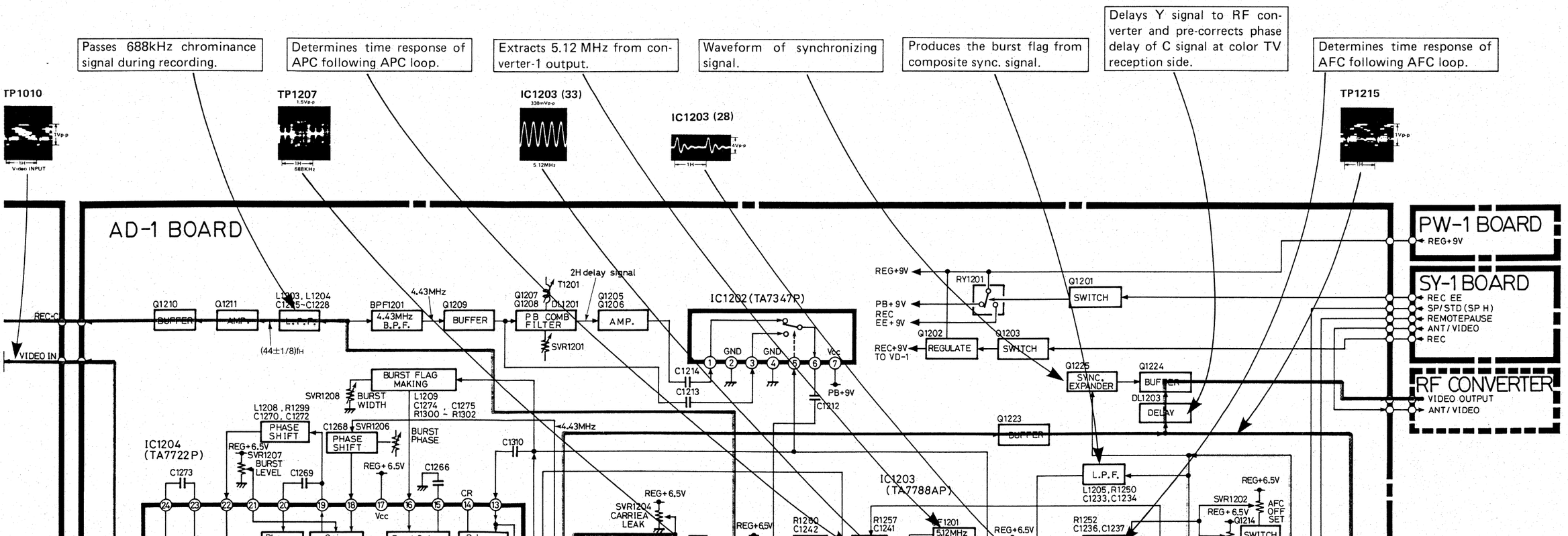
Emphasis of characteristic fluctuation (by level).

Separates signal fror video signal

JIT BLOCK DIAGRAM

DE SIGNAL FLOW

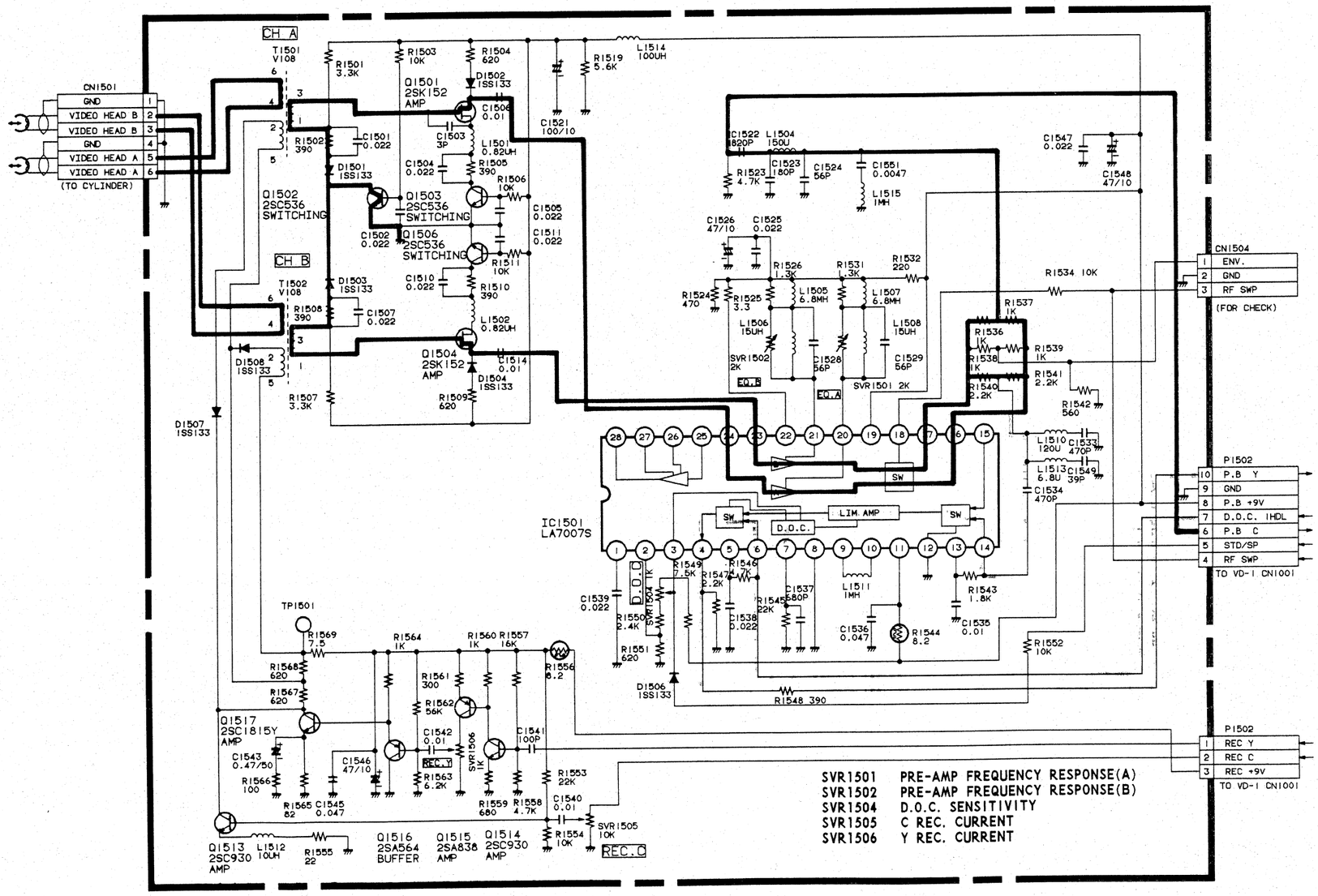
	CONTROL SIGNAL
	APC loop
ıal	ACC loop
ignal	AFC loop

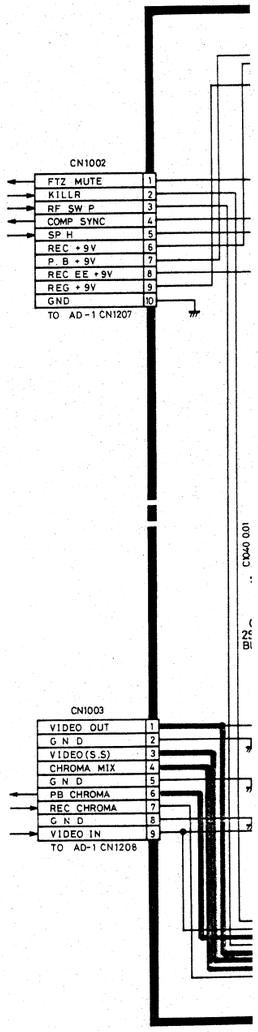


VIDEO IN
BUFFER
AMP.
L.P.F.
(44±1/8)fH
4.43MHz B.P.F.
BUFFER
PB COMB FILTER
SVR1201
AMP.
GND
GND
Vcc
C1214
C1213
PB+9V
C1212
EE+9V
Q1202
REGULATE
REC+9V TO VD-1
Q1203
SWITCH
Q1225
SYNC. EXPANDER
Q1224
BUFFER
DL1203
DELAY
Q1223
BUFFER
REMOTE PAUSE
ANT/VIDEO
REC
RF CONVERTER
VIDEO OUTPUT
ANT/VIDEO
BURST FLAG MAKING
SVR1208 BURST WIDTH
L1209 C1274 , C1275 R1300 ~ R1302
L1208 , R1299 C1270 , C1272
PHASE SHIFT
C1268 SVR1206
PHASE SHIFT
BURST PHASE
4.43MHz
C1310
IC1204 (TA7722P)
REG+6.5V
SVR1207 BURST LEVEL
C1273
C1269
REG+6.5V
C1266
CR
Vcc
Phase Invert
Gain Control Amp
Burst Gate Generation
Pulse Wavetorm
Jump DET.
Buffer
F.F.
Switch Control
Cleaning Flag Gen.
Buffer
DL Drive
Switch
VCO
AFC DET.
Gain ControlAmp
GND
N.C.
REG+6.5V
REG+6.5V
C1286
T1202 HORIZONTAL PHASE
DELAY
DL1202
SVR1211 1H GAIN
SVR 1210
C1283 SPECIAL PB VCO
C1280
AFC FILTER
R1304 , C1281 , C1282
IC1203 (TA7788AP)
REG+6.5V
SVR1204 CARRIEA LEAK
REG+6.5V
R1261
C1245
R1260 C1242
ACK FILTER
R1257 C1241
ARC FILTER
F1201 5.12MHz FILTER
REG+6.5V
L.P.F.
L1205 , R1250 C1233 , C1234
REG+6.5V
SVR1202
REG+6.5V
Q1214
AFC OFF SET
SWITCH
R1252 C1236 , C1237
AFC FILTER
SVR1203 VCO FREE OSC. FREQUENCY
VIDEO LINE OUT
REMOTE PAUSE
VIDEO LINE IN
Chroma mode
REC-Killer
NTSC
PAL
CONV.2
ACK
Killer
Killer DET.
Vcc2
GND2
5.12MHz
Phase Invert
PB·PAL
Pilot Burst Flag
Burst Flag1
Burst Flag 2
RF SW
1/2fH
CONV.1
Counter
VCO
AFC DET.
HD GEN.
REC
PB
5fH
1/2
1/2fH
Burst Flag1
Burst Emphasis
NTSC
Chroma Inverter
F.F.
Amp.
PB
REC
APC-ID
Burst ID DET.
Burst Flag2
SYNC DET.
(44±1/8)fH
4.43MHz
NTSC
PAL
Approximate 5.5MHz
Pilot Burst Flag
Pilot BurstADD
ACC Amp.
Channel Hold
Reference signal
fsc VXO
PB or SECAM
3fsc OSC.
REC·PAL
ACC DET.
REC
PB
GND
Vcc
C1263
R1292
SVR1205 PILOT BURST
REG+6.5V
R1286 C1254 C1256 C1258
ACC FILTER
C1253
PB+9V
R1248
C1249 R1277
PHASE SHIFT
R1247
REG+6.5V
N.C.
N.C.
REG+6.5V
R1273
IC1205 (TA7357AP)
Ripple Cansel
Sync. Separator
HD Generator
GND
N.C.
Vcc
Q1221
MODE SWITCH
Q1222
BUFFER
R1317
R1314
R1314
REG+9V
REG+9V
REG+9V
Q1217
ACC FILTER SWITCH
IC1201 (LA7016)
X1201 C1250
R1348 C1251
VC1201 VXO
VXO FREE REFERENCE
Q1216
SWITCH
BPF1202 4.43MHz B.P.F.
Q1218
BUFFER
4.43MHz
Q1215
BUFFER
Q1204
BUFFER
REC EE+9V
SP H
Composite Video signal
FTZ mute signal
Composite sync.(V and H) signal
Composite sync.
From AUDIO hi-fi CIRCUIT
CIRCUIT EXT/TU
REC EE+9V
B1201
SWITCH
TUNER AUDIO MUTE SIGNAL
9V
EXT+SIMUL
TUNER
0V
D1206
D1201
TN-2 BOARD
TUNER AUDIO MUTE
TN-1 BOARD
VIDEO IN
FTZ MUTE
D1202
SV-1 BOARD
COMPOSITE SYNC.
RF SWITCH PULSE
C·MIX
KILLER
C signal + DC level
PB·C
SP/STD
FTZ MUTE
OSITE SYNC.
VIDEO OUT
SW PULSE
VIDEO SS
IC1203 (1)
IC1203 (30)
4.43MHz
1Vp-p
3.5V
1H
TP1209
680mVp-p
4.43MHz
IC1203 (14)
600mVp-p
4.43MHz
IC1203 (21)
420mVp-p
5.5MHz
tic
Separates the synchronizing signal from the composite video signal.
Extracts chrominance signal from composite video signal.
Determines time response of ACC following ACC loop.
During playback becomes X'tal OSC. of reference frequency (f_{SC}); during recording becomes VXO and locks to sub-carrier frequency of input.
$-\pi/4$ phase shifter.
Improves time response of AFC and APC during switching by the RF switch pulse.
Mutes TUNER AUDIO signal when input selector set to EXT and SIMULCAST.

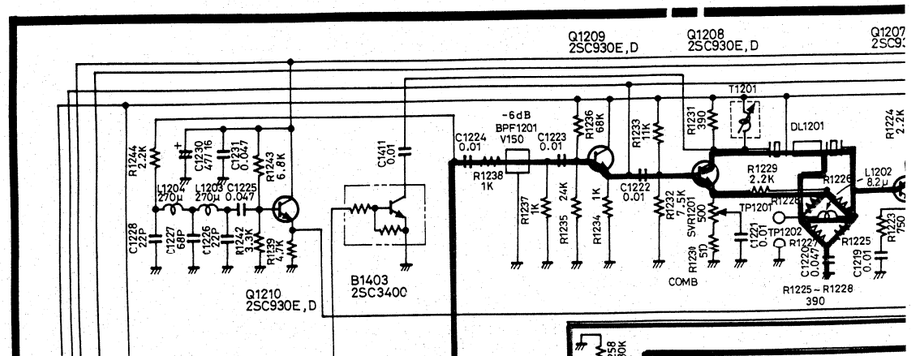
VIDEO CIRCUIT D

P.C.B. VP-1 VIDEO PREAMP CIRCUIT (143-0-230V40105)





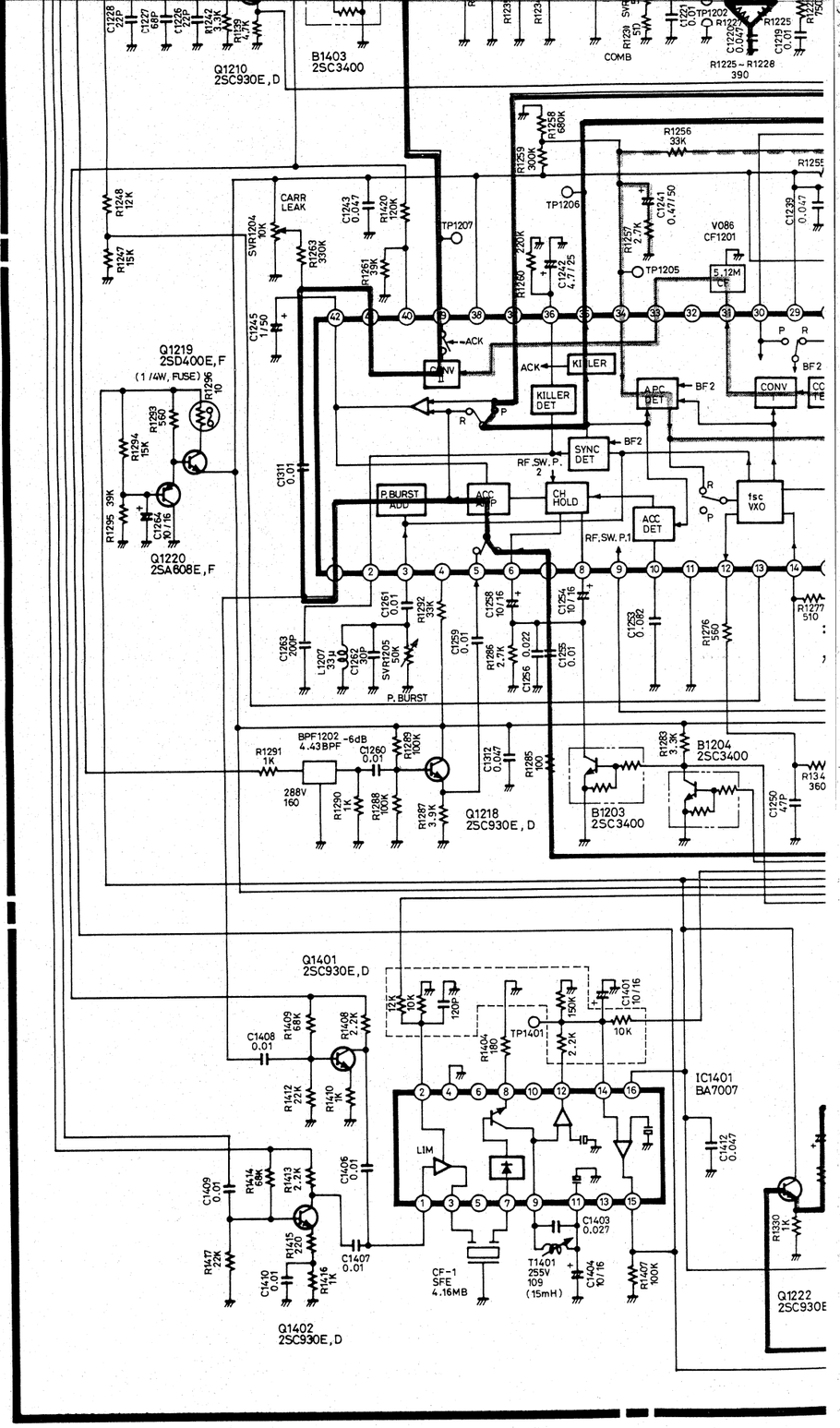
PLAY MODE SIGNAL FLOW



PLAY MODE SIGNAL FLOW

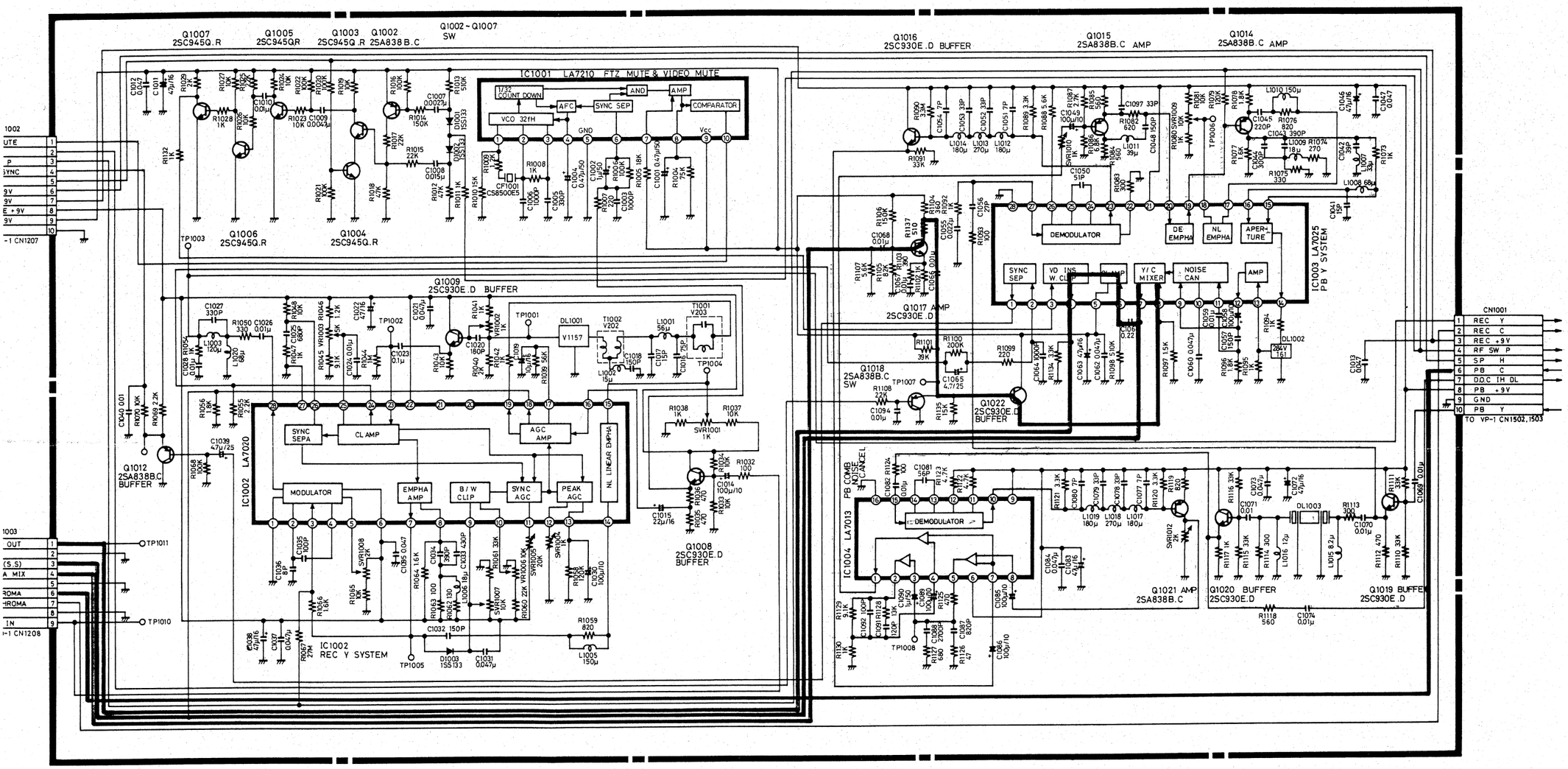
PLAY MODE MAIN SIGNAL
Video (Y/C) signal
Luminance (Y) signal
Chrominance (C) signal

CONTROL SIGNAL
APC loop
ACC loop
AFC loop (Special PB)
APC-ID loop

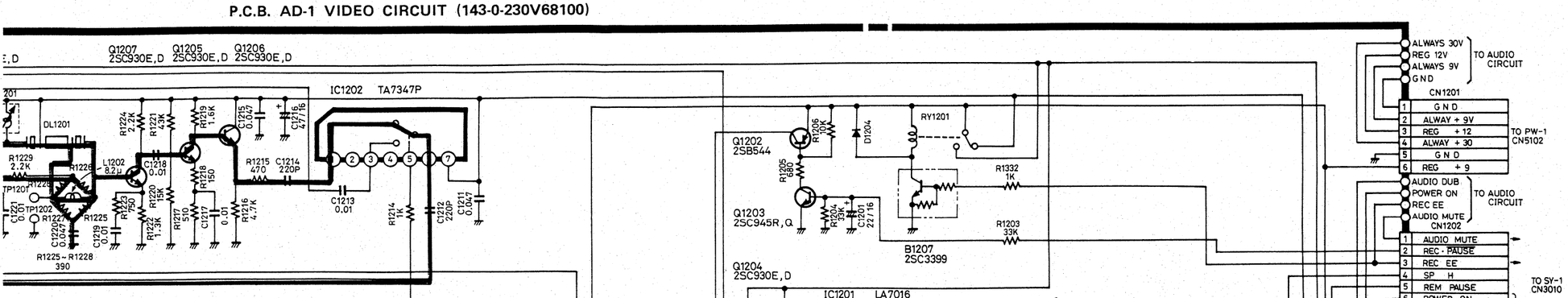


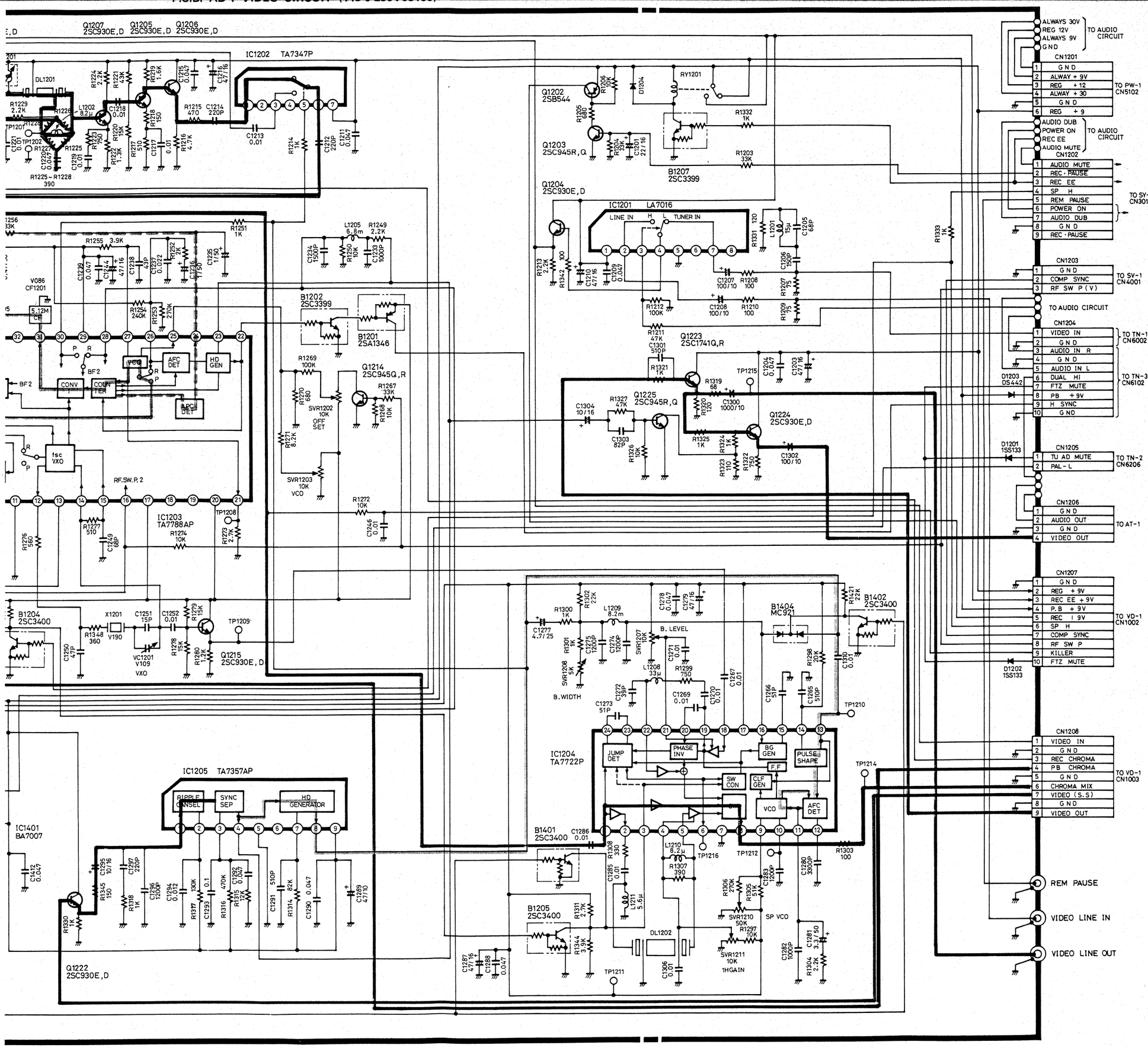
RCUIT DIAGRAM

P.C.B. VD-1 VIDEO CIRCUIT (143-0-230V57722)

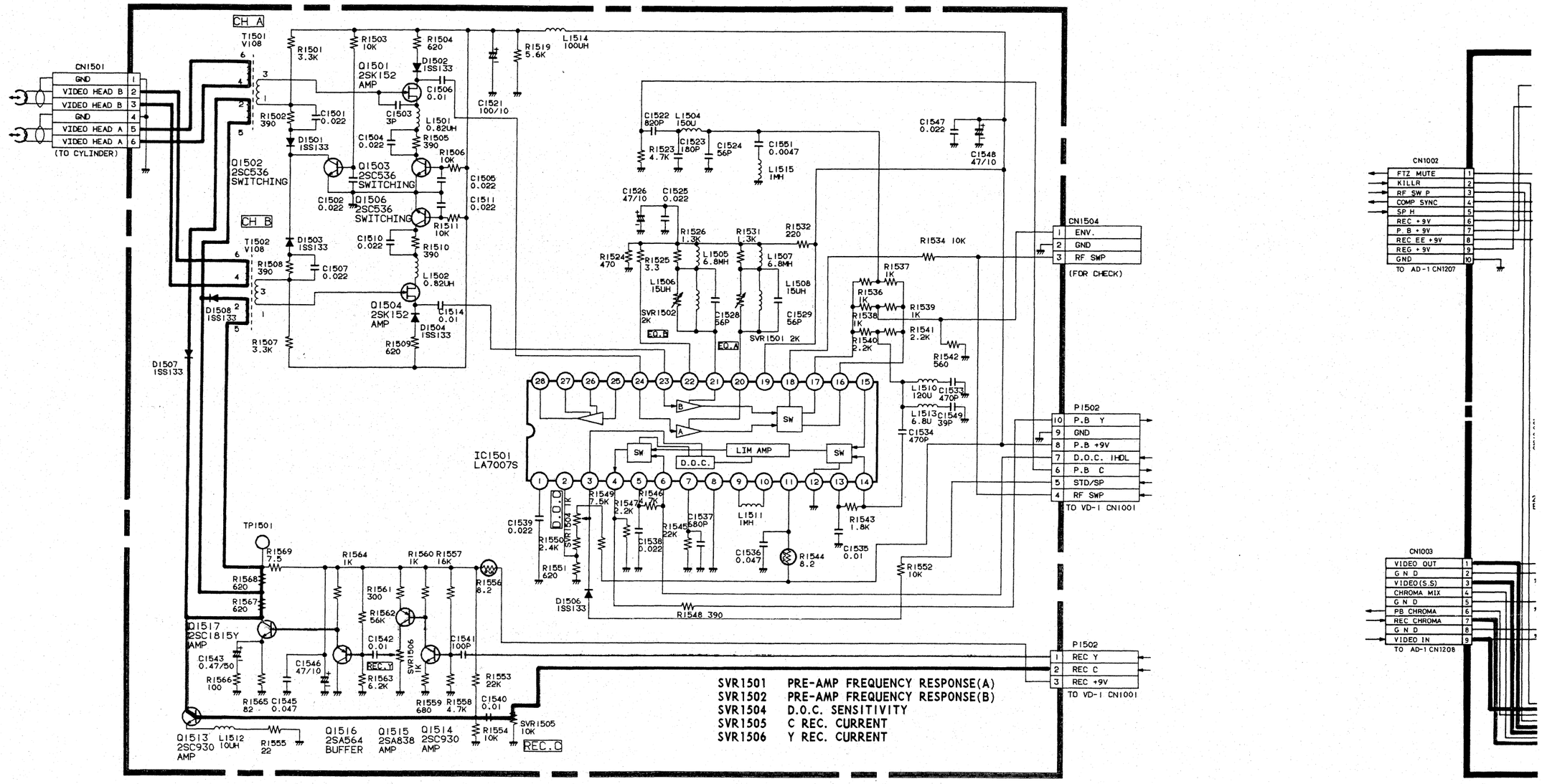


P.C.B. AD-1 VIDEO CIRCUIT (143-0-230V68100)

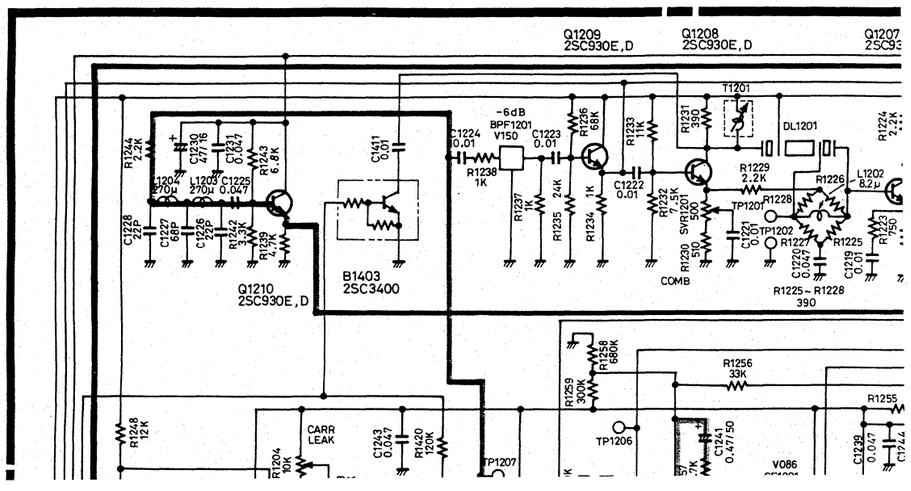



Q1207 2SC930E,D
Q1205 2SC930E,D
Q1206 2SC930E,D
IC1202 TA7347P
DL1201
Q1202 2SB544
Q1203 2SC945R,Q
RY1201
B1207 2SC3399
Q1204 2SC930E,D
IC1201 LA7016
LINE IN
TUNER IN
B1202 2SC3399
B1201 2SA1346
V086 CF1201
5.12M CF
IC1203 TA7788AP
RF.SW.P.2
fsc VXO
CONV
COUNTER
APC DET
VCO
AFC DET
HD GEN
SVR1202 10K OFF SET
SVR1203 10K VCO
Q1214 2SC945Q,R
Q1223 2SC1741Q,R
Q1225 2SC945R,Q
Q1224 2SC930E,D
B1204 2SC3400
X1201
V190
VC1201 V109 VXO
Q1215 2SC930E,D
IC1205 TA7357AP
RIPPLE CANSEL
SYNC SEP
HD GENERATOR
IC1401 BA7007
Q1222 2SC930E,D
B1205 2SC3400
B1401 2SC3400
B1402 2SC3400
B1404 MC921
IC1204 TA7722P
JUMP DET
PHASE INV
BG GEN
PULSE SHAPE
F.F
SW CON
CLP GEN
VCO
AFC DET
SP VCO
B. LEVEL
B.WIDTH
DL1202
ALWAYS 30V
REG 12V
ALWAY 9V
G N D
TO AUDIO CIRCUIT
CN1201
1 G N D
2 ALWAY + 9V
3 REG + 12
4 ALWAY + 30
5 G N D
6 REG + 9
TO PW-1 CN5102
AUDIO DUB
POWER ON
REC EE
AUDIO MUTE
TO AUDIO CIRCUIT
CN1202
1 AUDIO MUTE
2 REC · PAUSE
3 REC EE
4 SP H
5 REM PAUSE
6 POWER ON
7 AUDIO DUB
8 G N D
9 REC · PAUSE
TO SY-1 CN3010
CN1203
1 G N D
2 COMP SYNC
3 RF SW P (V)
TO SY-1 CN4001
TO AUDIO CIRCUIT
CN1204
1 VIDEO IN
2 G N D
3 AUDIO IN R
4 G N D
5 AUDIO IN L
6 DUAL Hi
7 FTZ MUTE
8 PB +9V
9 H SYNC
10 G ND
TO TN-1 CN6002
TO TN-3 CN6102
D1203 DS442
D1201 1SS133
CN1205
1 TU AD MUTE
2 PAL-L
TO TN-2 CN6206
CN1206
1 G N D
2 AUDIO OUT
3 G N D
4 VIDEO OUT
TO AT-1
CN1207
1 G N D
2 REG + 9V
3 REC EE + 9V
4 P.B + 9V
5 REC I 9V
6 SP H
7 COMP SYNC
8 RF SW P
9 KILLER
10 FTZ MUTE
TO VD-1 CN1002
D1202 1SS133
CN1208
1 VIDEO IN
2 G N D
3 REC CHROMA
4 PB CHROMA
5 G N D
6 CHROMA MIX
7 VIDEO (S.S)
8 G N D
9 VIDEO OUT
TO VD-1 CN1003
REM PAUSE
VIDEO LINE IN
VIDEO LINE OUT

P.C.B. VP-1 VIDEO PREAMP CIRCUIT (143-0-230V40105)



REC MODE SIGNAL FLOW

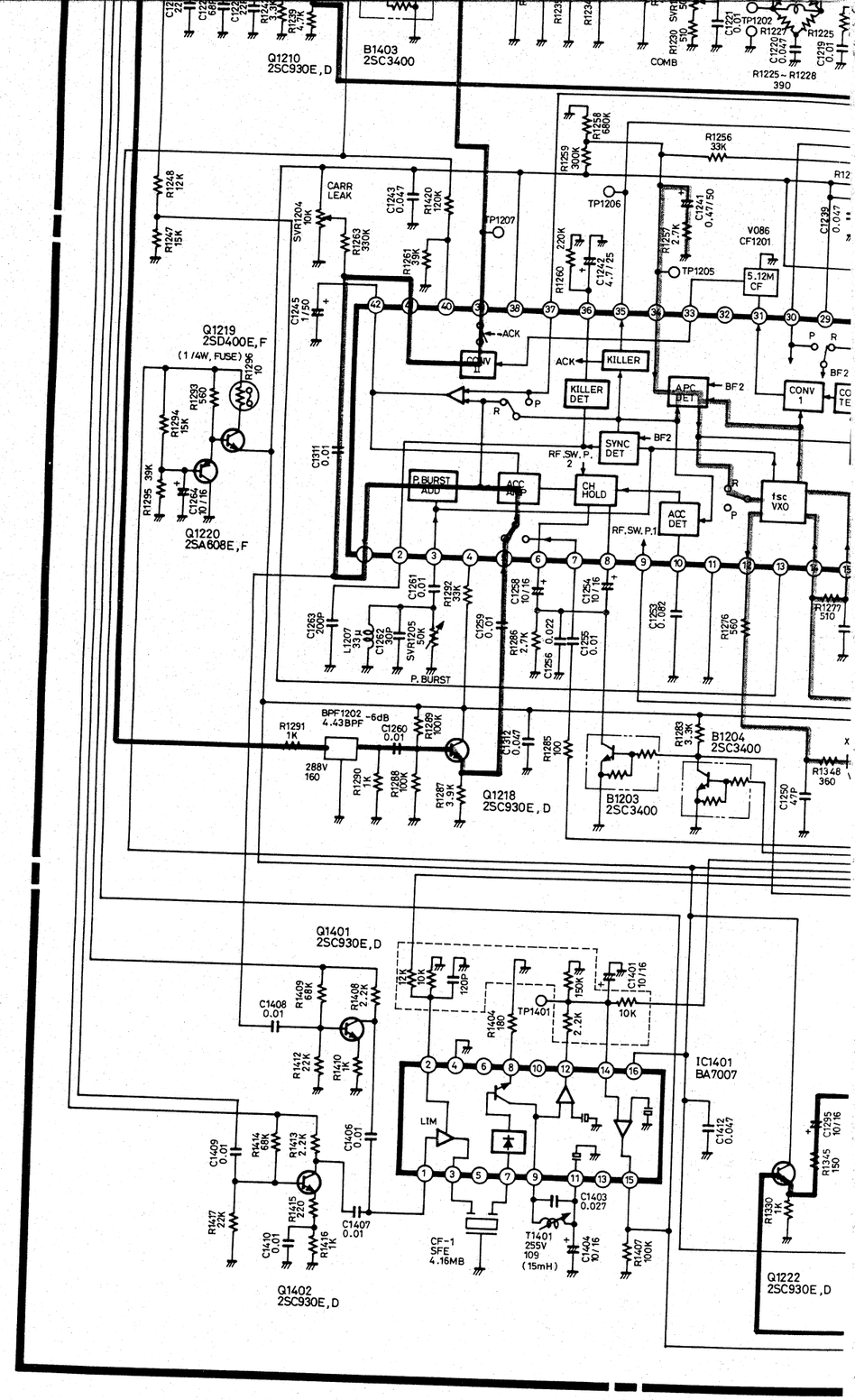


REC MODE MAINSIGNAL

REC MODE SIGNAL FLOW

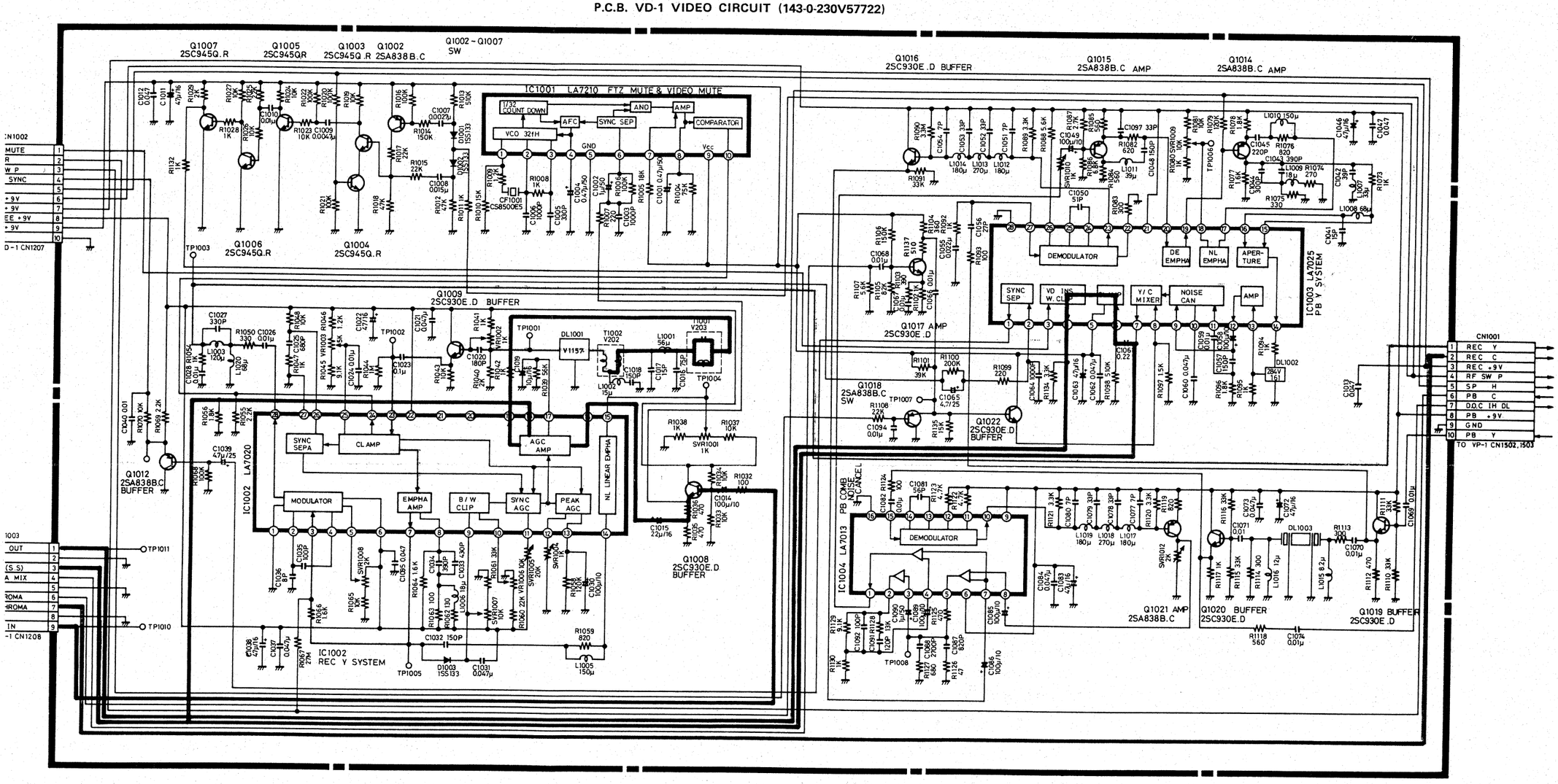
REC MODE MAINSIGNAL
——— Video (Y/C) signal
····· Luminance (Y) signal
——— Chrominance (C) signal

CONTROL SIGNAL
——— APC loop
ACC loop
——— AFC loop



IRCUIT DIAGRAM

P.C.B. VD-1 VIDEO CIRCUIT (143-0-230V57722)



P.C.B. AD-1 VIDEO CIRCUIT (143-0-230V68100)

IC1203 TA7788AP

IC1205 TA7357AP

IC1204 TA7722P

IC1201 LA7016

Q1202 2SB544

Q1203 2SC945R,Q

B1207 2SC3399

Q1204 2SC930E,D

Q1223 2SC1741Q,R

Q1225 2SC945R,Q

Q1224 2SC930E,D

B1202 2SC3399

B1201 2SA1346

Q1214 2SC945Q,R

B1204 2SC3400

Q1215 2SC930E,D

B1401 2SC3400

B1205 2SC3400

Q1222 2SC930E,D

B1404 MC921

B1402 2SC3400

CN5102 TO PW-1
- 3 REG +12
- 4 ALWAY +30
- 5 GND
- 6 REG +9

CN1202 TO AUDIO CIRCUIT
- AUDIO DUB
- POWER ON
- REC EE
- AUDIO MUTE

CN3010 TO SY-1
- 1 AUDIO MUTE
- 2 REC · PAUSE
- 3 REC EE
- 4 SP H
- 5 REM PAUSE
- 6 POWER ON
- 7 AUDIO DUB
- 8 GND
- 9 REC · PAUSE

CN4001 TO SV-1
- CN1203
- 1 GND
- 2 COMP SYNC
- 3 RF SW P (V)

TO AUDIO CIRCUIT

CN6002 TO TN-1
CN6102 TO TN-3
- CN1204
- 1 VIDEO IN
- 2 GND
- 3 AUDIO IN R
- 4 GND
- 5 AUDIO IN L
- 6 DUAL Hi
- 7 FTZ MUTE
- 8 PB +9V
- 9 H SYNC
- 10 GND

D1203 DS442

CN6206 TO TN-2
- CN1205
- 1 TU AD MUTE
- 2 PAL-L

D1201 1SS133

CN1206 TO AT-1
- 1 GND
- 2 AUDIO OUT
- 3 GND
- 4 VIDEO OUT

CN1002 TO VD-1
- CN1207
- 1 GND
- 2 REG +9V
- 3 REC EE +9V
- 4 P.B +9V
- 5 REC +9V
- 6 SP H
- 7 COMP SYNC
- 8 RF SW P
- 9 KILLER
- 10 FTZ MUTE

D1202 1SS133

CN1003 TO VD-1
- CN1208
- 1 VIDEO IN
- 2 GND
- 3 REC CHROMA
- 4 PB CHROMA
- 5 GND
- 6 CHROMA MIX
- 7 VIDEO (S.S)
- 8 GND
- 9 VIDEO OUT

REM PAUSE

VIDEO LINE IN

VIDEO LINE OUT